Surinder Singh
18/2/80

Chorosynth
a low-cost synthesiser

printer
for microprocessors
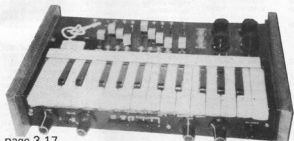
guitar with keyboard
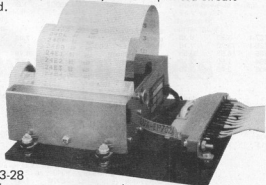### plays like a guitar—
sounds like an organ

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page 3-17
Although it may look, at first sight, like a toy piano or organ, the Chorosynth is actually a real instrument that has been used with success at live performances. As the 'hands down' winner of our Eurotronics competition, it certainly merits a printed circuit board.

page 3-28
As microprocessor systems become ever more like their big brothers — commercial computers — one of the features that must be added is a 'micro paper printout'. The printer for microprocessors described in this issue is a good example.

coming soon

in the April issue of Elektor . . .
electronics in cars in the Eighties!
with designs for electronic ignition systems, intelligent windscreen wiper delay, instant fuel consumption meter, various indicator circuits, and many more.

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Accidental destruction of recorded data on magnetic tape

Many myths surround the accidental erasure of magnetic media. This article, by the manufacturers of Scotch Instrumentation Tape, shows the results of an extensive test program to discover just how much tape will stand before loss of information occurs. Although the purpose of the tests was primarily with digital data, as used in the computer world, similar effects will also occur when storing analogue information on magnetic tape.

Magnetic fields

The obvious threat to magnetic media is the permanent magnet which can produce strong, steady magnetic erasing fields and needs no external power source for producing or maintaining its field.

A variety of tests were carried out using permanent magnets and it was found that a signal level loss of less than 50% of the original value did not result in a loss of data, because the data at this level can be still recovered during normal operations.

For instance, when a horseshoe magnet strong enough to lift 40 lbs (18 kg) was placed directly against the edge flange of the reel of tape, a signal reduction of 80% resulted at the beginning of the tape (See Fig. 1). This loss of signal decreased steadily along the tape from that maximum of 80% until at only 350 feet (107 metres) into the tape where the signal level was 50% of the original recorded level. At this point a tape can still be read.

The decrease in signal loss from 80% to 50% over 350 feet can be explained by the fact that the further into the tape the progressively weaker becomes the magnetic field and thus the erasing effects.

Since the intensity of a magnetic field falls off approximately as the cube of the distance from the magnet (at 10" away from the magnet its field is roughly 1/1,000 as strong as when it is 1" away), it was seen that when the horseshoe magnet was held 1" (25 mm) from the edge of the reel, a signal loss of only 22% occured: this reduction of signal did not result in any data loss at all. It would therefore be impossible for any concealable magnet to cause data loss if there is a distance of at least 6" between the magnetic source and the tape.

Even when a large scrap metal 'Electro' magnet was placed 1.3 feet (0.41 metre) above recorded tapes no data loss occured.

However, it must be pointed out that any small permanent magnet could cause a loss of data if placed directly next to or on to, the surface of the magnetic media.

As the magnetic media read/write head is also an electro-magnet, it is also quite capable of becoming permanently magnetized and thus cause some erasure and consequent data loss. This problem can be overcome by regular use of a head demagnetizing or degaussing device. As far as equipment such as transformers and electric motors are concerned, both of which produce magnetic fields, if this type of electrical equipment is enclosed within a cabinet allowing two or three inches of spacing between the magnetic tape and the electrical component there will be no danger of erasure.

Radar signals

Very often magnetic tapes are transported within areas where radar antenna are transmitting, for instance within airport locations, defence areas, aboard ships etc.

To gauge what effects, if any, radar signals had on magnetic tape, tests were carried out using irradiation by C-Band, L-Band and X-Band radar systems, ranging from 200,000 watts for the C- and L-Band and 500,000 watts for the X-Band, of peak power. The tape was placed at 100' (30.5 metres), 50' (15.2 metres) and 10' (3 metres) in the direct signal path of stationary antenna. No signal loss or data erasure was observed, and would probably not occur unless the tape was in actual direct contact with the antenna.

Metal detectors

Although there was, and probably still is, much concern over airport metal
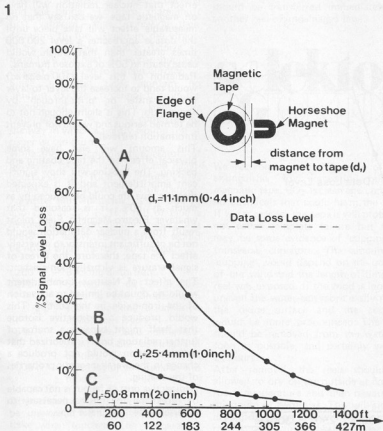
Figure 1. Permanent magnetic fields.

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detectors and the effect these instruments had on magnetic media, it has been shown that this concern is no longer necessary.

Many experiments were carried out using both the 'walk-through' type of detectors and the 'hand-held' units. There was no instance where data loss occured. Even when the highest intensity type of detectors using a rating of 398 A/m (5 Oe) was tried, no significant signal loss was seen as shown in figure 2. Today the maximum strength of a walk through metal detector is recommended at approx. 95 A/m (1.19 Oe), a level much too small to cause erasure of magnetic media.

X-Ray contamination

On occasions, magnetic media can be subjected to X-ray energy, especially by airport surveillance systems.

Various recorded magnetic media have been subjected to extremely high (lethal to humans) X-ray doses and no erasure or data loss occurred. This was the same result for magnetic tapes subjected to airport X-ray techniques.

High electrical charges

High voltages were generated by automobile ignition coils and they were applied directly across recorded magnetic tape. More than 15,000 volts were generated, and the ignition coils produced arcs which struck directly on to the tape. No erasure or data loss took place. Only if the voltage was so high that excessive arcing actually physically damaged the tape would data loss occur.

Micro-waves

Recorded cassettes were subjected to the maximum microwave energy, in fact until they became warm, when placed

inside a normal microwave oven. When the cassettes wer re-read, there was no measurable data loss. This would include any leakage areas existing around ovens.

Televisions

Because television receivers are sources of magnetic fields, high voltage and X-rays, various tests were performed by placing recorded magnetic tape both inside and outside of different receivers, including colour sets. Despite the high voltage potential and X-ray sources, no signal or data loss was recorded.

However, as receivers have many magnetic field producing components, and many receivers were not tested, it is advised not to store magnetic media on top of these devices.

Nuclear radiation

A gamma ray dose of approximately 3.0 megarads was directed at a recorded digital tape cassette while in a gamma ray pool for at least 1½ hours. No data loss was recorded at this level.

However, as a general statement on the effect that nuclear radiation will have on magnetic tape, we can say that no measurable effect will take place until the dosage approaches a level 200,000 times greater than that which would cause death in 50% of exposed humans. Radiation of this level (100 megarad) would tend to increase the layer-to-layer signal transfer or 'print-through' by about 4 db. This is slight enough not to be termed serious and would not prevent information retrieval.

This amount will also have some physical effect on the tape coating and backing. The backing will show significant embrittlement and it is expected that the tape life could be reduced by as much as 60%. It is reasoned that whatever Electro-Magnetic Field might result from a nuclear detonation would not be of sufficient intensity to adversely affect the tape; therefore, the threat of signal erasure is virtually non-existent.

The effect of Neutron bombardment would no doubt be limited to activation of the iron-oxide in the coating. This would produce a radioactive isotope that itself might become a source of further radiation, but it is theorized that such activation would not produce a change in the overall magnetic properties of the coating.

Radioactive dust or fallout is not capable of producing the dosage necessary to adversely affect magnetic tape.

2

Figure 2. Electromagnetic fields.

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Laser beams
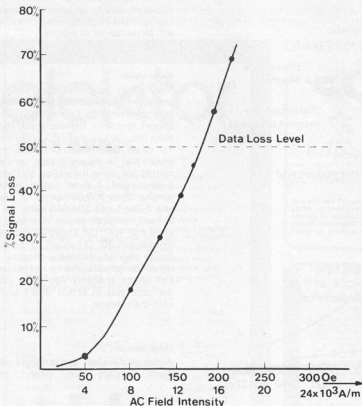
Laser beams are sometimes used as a recording medium, i.e. heat or 'thermoremanent' recording and the 'medium burning mode'.

However, to prove that data destruction from a distance is not practical, a recorded magnetic tape was subjected to a laser beam spread over a wide area of the tape. No data loss occurred. Only if a very high intensity beam were to be directed at the medium would there be sufficient heat to destroy the actual physical properties of the tape.

Effect of heat

For a substance to burn, there must be a breakdown of the organic materials contained in it. The organic materials in magnetic tape are the plastic backing and the binder. To burn, these must first vaporize — thus increasing their exposure to the oxygen in the atmosphere — and then rapidly oxidize to form light and heat. An ample supply of oxygen is required to sustain burning. Since magnetic tapes contain no 'built-in' oxidizer, it cannot burn in the absence of air. Simply stated, its behaviour can be closely compared to the way in which a tightly wound roll of paper would burn.

While the 'self-ignition' temperature of polyester backed tape is in the neighbourhood of $1000°F$, temperatures below that point can still cause damage. Polyester film will shrink $1\frac{1}{2}\%$ at $300°F$, and 25% at $325°F$. If a roll of tape is heated to the approximate temperatures presented below, certain effects would be noted when the roll had cooled.

$250°F$ —	Backing distortion.
$320°F$ —	Softening of both the backing and binder with some 'blocking' or adhesion of adjacent layers.
$550°F$ —	Darkening and embrittlement of the backing and binder.
$1000°F$ —	Charring of the backing and binder.

When charring occurs, the tape cannot be unwound from the reel as it will flake when touched. The temperature limitation of present day tapes is a function of the gamma ferric oxide pigment.

Two reels of computer magnetic tape functioned adequately after exposure to $210°F$ ($99°C$) and then refused to load into the tape transport after they had been subjected to $230°F$ ($110°C$). At

first, this appeared to be a permanent condition since these tapes had not recovered after six weeks in storage. However, after two months, they began to run again; the data could be recovered, but the oxide appeared to be shedding badly. In such cases, if possible, the data should be copied as soon as possible onto new tapes and the shedding tape discarded.

Theoretically, gamma ferric oxide particles which are the prevalent magnetic constituents used in most magnetic storage media, are capable of retaining data up to a temperature of approximately $1247°F$ ($675°C$). This temperature is known as the Curie temperature. On the other hand, chromium dioxide particles, used in audio and video tapes, have a Curie temperature of approximately $275°F$ ($135°C$) and are therefore far more susceptible to a heat related data loss.

Winding and storing magnetic tape properly will lessen the possibility of damage in the event of fire as tape is a poor conductor of heat. It is sometimes possible to recover information from a tape receiving slight fire damage by carefully rewinding it at minimum tension. The information it contains should be transferred immediately to another reel of undamaged tape.

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We recommend the CO_2 type of fire extinguisher for combating burning magnetic tape. CO_2 is clean and contains no chemicals that could harm the tape. If water reaches the tape it will probably not cause complete failure but there may be some evidence of 'cupping' or transverse curvature. The amount of 'cupping' would depend on the quality of the wind and the length of time the reel was exposed. If the wind is loose or uneven the water can more easily reach the oxide surface and the cupping would be more pronounced. The tape should be removed from the water as soon as possible, and certainly within 24 hours.

After removal, the reels should be allowed to dry on the outside at normal room temperature and then rewound a minimum of twotimes. This will aid the drying operation and will also help the reels to return to equilibrium faster.

If a temperature increase is also incurred while the tape is water soaked, steam or at least high humidity will be present. This is more likely to cause damage than water alone. A temperature in excess of $130°F$, with a relative humidity above 85% may cause layer to layer adhesion as well as some physical distortion.

Effect of low temperatures

Recorded computer tapes which were kept at a temperature of $-60°F$ ($-51°C$) for 24 hours, showed no data loss after being dried and relaxed — a process of careful, slow unwinding and rewinding.

In theory, no loss of magnetization should occur with magnetic oxide particles even in temperatures which are well below their Curie temperature although in most cases they will become stiff in nature and wet with condensed moisture.

It is important that after tapes have been kept in very low temperatures for long periods of time that they should be dried and relaxed over a period of days, at gradually increasing temperatures. This will relieve some of the stress which will have built up and will minimise the shrinkage that will have taken place.

Tapes that are cycled between temperature and humidity extremes can develop severe stresses within the reel due to the pressures caused by the plastic flow of the tape material. These stresses can lead to damaged regions on the tapes.

Conclusions

Failure of magnetic tapes as a reliable storage media is almost always due to some form of physical deterioration of the tape and not to the deterioration of the recorded data.

Magnetic fields were found to be the only type of energy that could damage the recorded data without actually physically affecting the magnetic media. Only a few inches of spacing is quite sufficient to prevent erasure of data by any concealable magnet which could feasibly be found in a normal environment. It has been determined that normally there is no need to shield the stored data against X-rays, high voltage fields, nuclear radiation, high frequency fields or light energy.
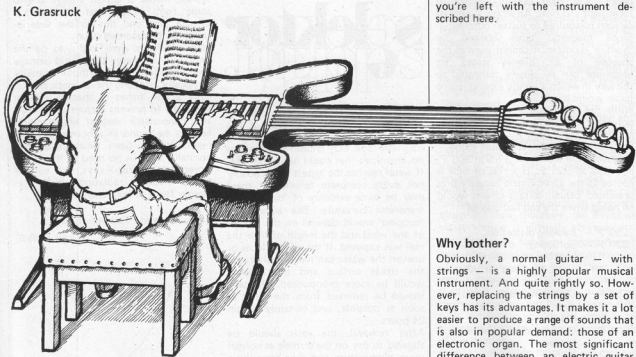
Instrumentation Talk volume 2.

(529 S)

selektor

clavitar

when is a guitar not a guitar?

when it has keys!

K. Grasruck

This is a new musical instrument. You might not notice it, at first sight, because it looks a bit like an electric guitar. You might not notice it at first 'hearing', either, because it can sound a bit like an electronic organ. It is played in approximately the same way as a guitar, so that guitarists should be able to get the hang of it quite quickly. They can then produce the full, rich sounds of an organ by using a finger-board instead of a keyboard.

A guitar has six strings — normally speaking. Each string is tuned to a specific note which can be considered as the 'basic note' for that particular string.

When playing it, the fingers of one hand press one or more of the strings down onto the 'frets' along the neck of the guitar. This reduces the effective length of the string, so that it produces a higher note when it is struck. Guitars are constructed in such a way that each successive fret corresponds to a half-note increase. By depressing several strings at once it is possible to play complete chords. There are a large number of standard chords for guitars, each with its own characteristic finger positions.

No strings attached

Now for something completely different. Instead of pushing down (or 'stopping') the strings at the right places, push down on keys! Each key can be arranged to correspond to a specific note; then, provided they are positioned correctly 'between the frets', the normal guitar finger positions will select the same chords.

If a further small group of keys is used to 'pluck the strings', there is no further need for the original strings; they only get in the way. Take them off, and you're left with the instrument described here.

Why bother?

Obviously, a normal guitar — with strings — is a highly popular musical instrument. And quite rightly so. However, replacing the strings by a set of keys has its advantages. It makes it a lot easier to produce a range of sounds that is also in popular demand: those of an electronic organ. The most significant difference between an electric guitar and the 'no-string' version described here is that, with a normal guitar, you have to 'keep it going'; with the keyed version, notes can be 'held' for as long as you like — as with an organ. On the other hand, the percussive sound of a normal guitar can easily be made, electronically, when using the no-string version. Since this 'decay' option can be switched on or off as desired, the result is a highly flexible instrument: from

organ to guitar at the touch of a switch. The guitar-with-keys can even be played single-handed — literally. After all, only one hand determines the chords to be played, and that's all that the electronics need to know. If the 'left-hand' keys are bridged, a note will sound as soon as it is selected. The instrument can then be played with one hand!

What it looks like

For reasons of cost and for ease of construction, some of the possibilities of a 'real' guitar were omitted. Quite apart from the fact that you can't 'pluck' a pushbutton, there are three further simplifications:

- Instead of six strings along the 'neck' of the instrument, there are only four. The two lower strings are omitted, leaving E', B', G and D.
- Somewhat surprisingly, the other hand can 'strike' five strings. The fifth is added electronically. It is lower than the others, corresponding to an A string; it produces a note that is one octave lower than that selected for either the B' or the G string. Normally the musician must deliberately select this note by pressing down on the fifth string at the correct point between the frets.
- By far the most drastic simplification is the reduced number of frets: only five are used. A normal guitar will have more than twelve — but in practice, you can do quite a lot with only five. Basically, it means that you are limited to playing in the lowest positions. However, it also means that only twenty

keys are needed along the 'neck' of the instrument — instead of the large number that would be required for a full-scale imitation of a guitar.

How it works . . .

A block diagram is given in figure 1; at the same time, this drawing gives some idea of the shape and key positioning for the instrument. The twenty keys that determine the notes for the four 'strings' are mounted at the end of the 'neck'. These we can refer to as the 'note keys'. In the circuits, they are labelled according to the note that they produce: 'S_d' say, corresponds to a D. In some cases, where more than one key produces the same note, these will be indicated as $S_{c'1}$ and $S_{c'2}$. The lowest C is given as c, one octave higher is c' and two octaves higher is c''.

S_A, S_D, S_G, $S_{B'}$ and $S_{E'}$ are the five keys for 'striking the strings'; we'll call them the 'string keys'. Operating $S_{B'}$, for instance, produces the note selected (by $S_{e'1} \ldots S_{c'2}$) for the second string. S_A produces the note for the 'fifth string', as determined by the note keys of either the B' or G string. If none of the note keys are operated, the string keys will produce the open note for the corresponding string — as with a normal guitar.

For 'single-handed' playing this function can be switched off, so that no tone can sound until one or more of the note keys is operated. The string keys can then be bypassed. The instrument will now play in the same way as if all five string keys were held down continu-

ously: a note will sound as soon as a 'note key' is operated.

. . . electronically

The basic principles of the circuit can be derived from the same block diagram. The note keys are arranged in four groups of five keys; each group determines the frequency of a corresponding oscillator.

As required, the outputs from these oscillators can be passed through frequency divider stages, bringing the note down one or more octaves. This helps to counteract some of the disadvantages of the simplifications described earlier. The next step is a multiple mixer stage; among other things, this derives the note for the simulated fifth string from those produced by the second and third strings.

The string keys $S_A \ldots S_{E'}$ each drive an 'envelope generator'; these determine the output levels (including attack and decay) for each string, by controlling associated 'voltage controlled amplifiers' (VCAs).

The circuit

The complete circuit is quite a complicated affair — not the kind of thing that you knock together in an hour or two. A large number of (inexpensive) components are needed; there is a lot of wiring to and from switches; and there's quite a bit of tuning to do.

For clarity, the circuit has been broken up into six partial circuits. The note keys and the four oscillators are given in
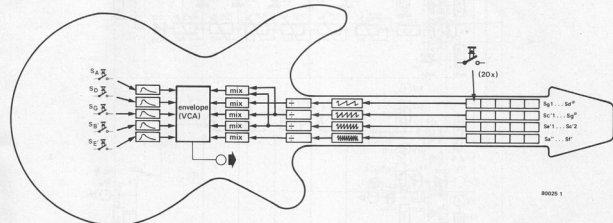
1

Figure 1. Block diagram of the complete instrument.

2

Figure 2. The circuit for the 'note keys' consists of four oscillators with associated presets and keys to select the desired notes.

3

Figure 3. The two types of divider stage. The simpler version (figure 3a) is used for the E' and D strings; the circuit given in figure 3b is for the other two.
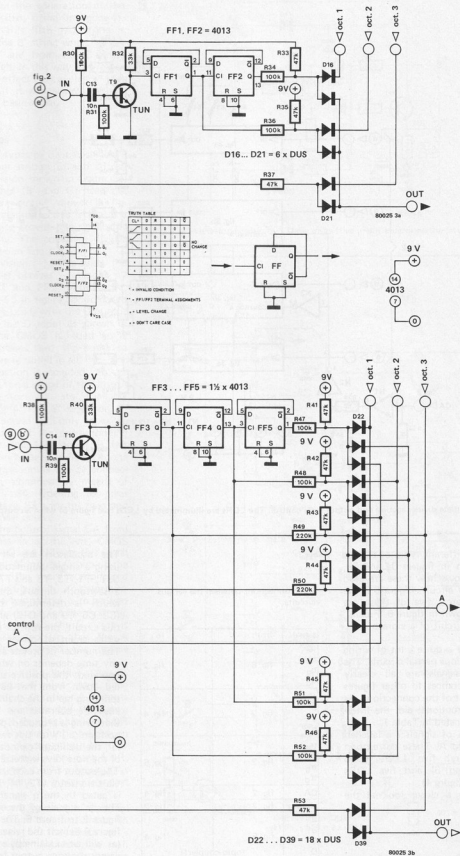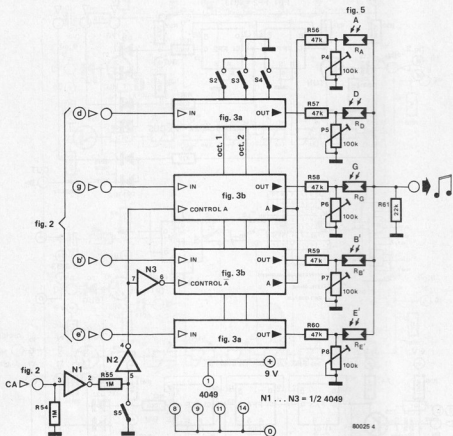
4

Figure 4. The complete divider section and output level control. The LDRs are illuminated by LEDs (see figure 5) when an output signal must be produced.

figure 2. Two different types of divider stage are shown in figures 3a and 3b, and figure 4 shows how these are used in the complete circuit. Figure 5 is the envelope generator; five of these are combined as shown in figure 6. Finally, figure 7 is the circuit of a vibrato oscillator.

Obviously, there is quite a lot of wiring between the various partial circuits. The corresponding signals are all clearly shown, and references to other figures are given at most of the connections. As a further constructional aid, the interconnections are listed in Table 1.

A special group of signals are labelled E', B', G, D and A. These correspond to opto-couplers: five LEDs (D45 in figures 5 and 6) and five LDRs (RE' . . . RA in figure 4).

Now, let's take a closer look at the various circuits.

The note keys

The circuit that produces the basic notes for the four strings is given in figure 2.

Table 1.

Interconnections between the various circuits.

signal	from	to
qe' qb' qg qd	fig. 6	fig. 2
e' b' g d	fig. 2	fig. 4
tre' trb' trg trd	fig. 2	fig. 6
CA	fig. 2	fig. 4
vibrato	fig. 7	fig. 2
E' B' G D A	fig. 5	fig. 4
	(opto-coupler!)	

The oscillators are very simple, each using a single unijunction transistor, or UJT: T1, T3, T5 and T7. They produce a sawtooth output, the frequency of which is determined by a capacitor (C3, C6, C9 and C12) and a resistor. In this circuit, the resistor is replaced by a series chain of preset potentiometers. The number of presets actually used at any time depends on which note key is operated: the positive supply voltage is fed down from that key to the corresponding tap in the chain. If none of the keys are operated the 'open note' for that string is produced, provided S1 is in position 'a'. With this switch in position 'b', the oscillator cannot run unless one of the note keys is pressed.

The output from each oscillator is taken via a transistor (T2, T4, T6 and T8); this is 'gated' by the 'q' signals from figure 6. The 'tr' signals go the other way, from figure 2 to figure 6. The circuit given in figure 6 detects the release of a note key (as will be explained), and then rapidly damps the tone output from that string. The circuit for the note keys that correspond to the B' string is more complicated than the other three, since it must

also provide the signal CA. This signal is used to control the generation of the sound for the fifth string (in figure 4). The note for this fifth (A) string is derived from the B' string when the CA signal is low, and from the G string when CA is high. In this way, the rather monotonous effect is avoided that would occur if it was always derived from the same basic signal.

Divider stages

Two different types of divider stage are used. The outer strings (E' and D) use the simple version in figure 3a; the inner strings (B' and G) need the more extensive circuit shown in figure 3b. The reason for this is that the inner strings also provide the signals for the fifth string. The complete divider section is shown in figure 4.

Frequency division — to provide the notes for lower octaves — is done by flip-flops FF1 and FF2 in figure 3a, and FF3 . . . FF5 in figure 3b. So-called D flip-flops are used, with the Q output connected to the D input as shown in figure 3a. The CMOS IC used (type CD 4013) contains two flip-flops, so that five ICs are required in all.

This divider section is included in order to extend the tonal range of the instrument. Three octaves are available (similar to 8', 4' or 2' stops on an organ); each 'octave' will only appear in the final output if the corresponding control input (oct 1 . . . oct 3) is left floating. As can be seen in figure 4, one or more of these inputs can be connected to supply common by means of switches S2 . . . S4. Opening S2 gives the lowest octave, S3 is for the next and S4 for the highest octave.

In figure 4, the control signal CA from figure 2 comes in at the left. CMOS inverters N1, N2 and N3 pass this signal to the 'control A' inputs of two of the divider stages (the more complicated version given in figure 3b), to determine which of these two strings is to be used for deriving the signal for the A string. If switch S5 is closed, the G string is permanently selected (so that the A string will always sound one octave lower).

The outputs from the divider stages go to a 'light-controlled' mixer stage. Each output (including the two A outputs for the fifth string) is fed through a resistor network that includes an LDR RE' . . . RA). The amount of light on the LDR determines the output level for that string. This means that each LDR, with an associated LED in figure 5, is basically equivalent to a very low-cost VCA.

The output from figure 4 is the final audio output from the instrument. In principle, it can be fed direct to a (power) amplifier; however, it will usually be far more useful to include a control amplifier with more-or-less extensive filtering capabilities — similar to those used in electronic organs.
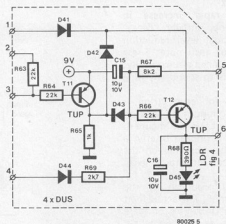

Figure 5. The envelope generator. These units (five in all) determine the 'attack' and 'decay' characteristics.
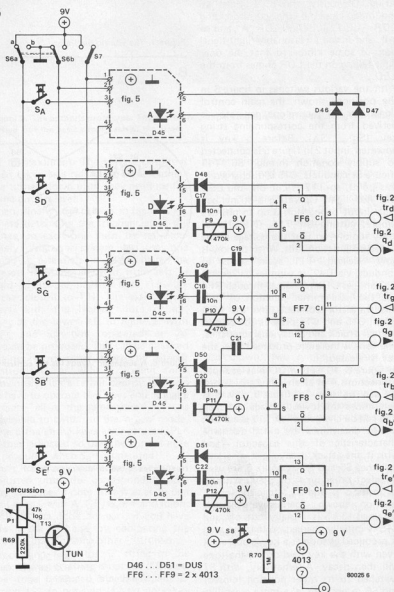

Figure 6. The complete envelope generator section. Four different ways of playing the instrument can be selected by means of S6 and S7.

Attack and decay

'Attack' is the rapid increase in amplitude of a tone, initially; 'decay' refers to the gradual level reduction towards the end of each note. These effects must be produced electronically, if this instrument is to sound anything like a guitar.

The output level for each string is determined by the amount of light on the corresponding LDR in figure 4. When a string key is pressed briefly, a short pulse of light must fall on the LDR, with the correct attack-decay characteristic.

The light pulse is produced by the circuit given in figure 5. The actual light source is the LED, D45. Five of these envelope generators, as they are called, are used in the complete instrument; the combined circuit is given in figure 6. One envelope generator is used for each string. Obviously, the LEDs must be mounted close to the corresponding LDRs (LED 45 beside LDR 'A', and so on), and inside a reasonably 'light-tight' box of some kind — so that the only light falling on the LDR comes from the LED.

With the various switches in figure 6 in the positions shown, the main control signal for each envelope generator is derived from the corresponding string key ($S_E' \ldots S_A$). Before a key is operated, input 2 in figure 5 is connected to supply common through S6. T11 (figure 5) conducts; C15 is discharged at this point, so T12 is cut off; the LED will not light. Operating a string key pulls input connection 3 up to supply supply, cutting off T11. C15 charges fairly rapidly, so that the voltage at the C15/R66 junction falls. When the key is now released, T11 again starts to conduct; via D42, it supplies current to T12 and, as C15 discharges through R66 and the base-emitter junction of T12, this transistor conducts and the LED lights. C15 and C16 provide the desired gradual attack and decay characteristics. The tone is therefore produced after the key is released.

In figure 6, P1 sets a DC bias at input connections 4 of all envelope generators. As can be seen in figure 5, this bias determines the lowest voltage to which C15 can be charged when T11 is cut off; it also determines the initial discharge characteristic of this capacitor — and with it, the attack/decay.

Switches S6 and S7 in figure 6 are used to select other types of 'play'. With S6 in position 'b' and S7 left in the (open) position shown, 'percussive' play is obtained: T11 is initially cut off and C15 is initially charged, so that the tone is produced as soon as a key is operated. Even with the key held down, the tone will then decay. Alternatively, with S7 switched to its other position (closed) and S6 in position 'a' a more organ-like effect is obtained: the note sounds as long as the string key is held down; after the key is released, the tone decays gradually ('sustain'). Finally, with S6 in
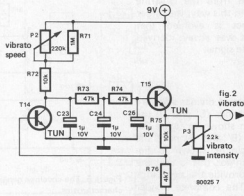

Figure 7. The vibrato oscillator.

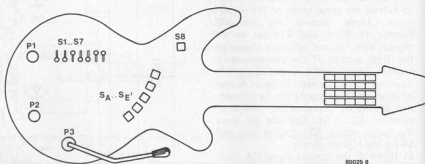

Figure 8. A suggested layout for the controls and keys.

Table 2

frequencies of the various notes (Hz)

	D string		G string		B' string		E' string	
	g	784.0	c'	1046.5	e'	1318.5	a'	1760.0
	f#	740.0	b'	987.8	d#'	1244.5	g#'	1661.2
	f	698.5	a#'	932.3	d'	1174.7	g'	1569.0
	e	659.3	a'	880.0	c#'	1108.7	f#'	1480.0
	d#	622.3	g#	830.6	c'	1046.5	f'	1396.9
open note:	d	587.3	g	784.0	b'	987.8	e'	1318.5

position 'b' and S7 closed, a tone will sound as soon as a note key is operated: the string keys need not be operated (actually, they will interrupt the tone — but the intention is to leave them alone . . .).

Four flip-flops, FF6 . . . FF9 in figure 6, ensure that the tone dies out rapidly when the corresponding note key is released — once again, as in a normal guitar. As mentioned earlier, they block the output in figure 2; furthermore, the Q output is used to discharge C15 (figure 5) rapidly. A 'damper' key is also provided: operating S8 mutes all strings rapidly — the same effect as placing the flat of the hand across the strings on a real guitar.

Vibrato

So far, nothing has been said about the 'vibrato' input to figure 2. It is connected to the circuit given in figure 7: a conventional very low frequency oscillator. Its output modulates the output frequency of the 'string' oscillators. The modulation depth — i.e. the extent to which the effect is audible — is varied by means of P3. The vibrato frequency is set by P2.

The circuit is designed so that the oscillator will stop when P2 is set to maximum resistance. P3 can then be used to introduce 'manual' vibrato — like the vibrato 'handle' on an electric guitar.

Construction and tuning

Figure 8 gives a suggested layout for the various keys and controls. Obviously, the positions can be modified according to personal taste; in particular, the 'damper' button (S8) might be mounted some way to the left of the string keys, so that it can be operated by the palm of the hand.

Good quality pushbutton keys should be used — they must operate easily, reliably and very often . . . As shown in the photo of the prototype (figure 9) the original author used 'Digitast' switches. He also extended the principle, building a complete six-string version.

As far as the electrical construction is concerned, the only 'fiddly bit' is the LED-LDR combination. Each pair must be adequately 'screened' from the other pairs and from ambient light.

A 9 V stabilised supply should be used. The total current consumption is only 150 mA.

Calibration is quite a job — as with most musical instruments. First: figure 6. The wipers of presets P9 . . . P12 are first turned down to supply common. They are then turned up to the point where a tone is just heard when the corresponding string key is operated (with S6 and S7 in the positions shown).

The presets in figure 4 (P4 . . . P8) are used to set the output levels of the strings. The best way is to first set them
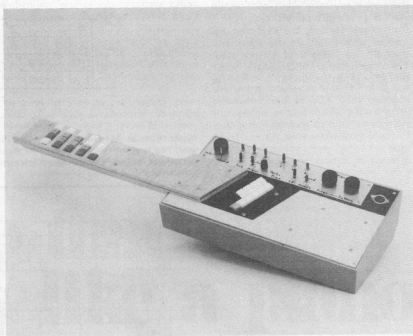

Figure 9. The author's prototype. This is a more complicated version, with six strings instead of five, but the basic circuits are all the same.

all to maximum. The softest output is then taken as a reference, and all others are turned down until they produce the same output level.

Then, finally, the presets in figure 2 . . . These determine the frequencies of the notes, so each one must be set accurately. Whatever system you use, one thing is essential: in each chain, the presets must be adjusted from left to right (first the highest note, i.e. the preset closest to T1). The easiest is to start with the highest note of all: $P_{a''}$, at the lower left in figure 2. With $S_{a''}$ and $S_{E'}$ both operated, an 'A' should be produced. This can be checked against some other (correctly tuned) instrument, or with a tuning fork. Obviously, it's a rather clumsy business to have to operate two keys and adjust a preset — all at the same time — so it is advisable to switch to the 'organ' mode, where the string keys are not used. $P_{a''}$ is then adjusted so that the correct note is produced when $S_{a''}$ is operated.

Each following preset in the chain is adjusted until the corresponding note key produces a note that is exactly one half-note lower: A, G#, G, F# and so on. This can be done 'by ear', if you have the knack, or else by comparing the note to that on some other instrument.

Proceeding from the E' string to the (next lower) B' string is not too difficult: the tone produced by the 'open' E' string (S1 in position 'a', no note key operated) is the same as that produced by the B' string when the highest note key ($S_{e'}$) is operated. After $P_{e'}$ has been correctly adjusted, that is. The other

presets in the B' chain can then be adjusted in half-note steps, as before.

The first preset in the G string chain ($P_{c''1}$) is adjusted to give the same note as the last key in the B' string ($S_{c''2}$). Once again, the remaining presets are tuned to half-note intervals. The open note of the G string is then used as reference for tuning the first preset in the D chain.

Along the way, some further checks are possible. This is illustrated in figure 10. Two keys with the same symbols should produce the same note; the open note for each string is shown to the right of the keys. Some octave relationships are also shown: two symbols in one key indicates that the note is one octave higher than that produced by the key with only one symbol. For example, the extreme left-hand note key on the upper string (S_{g1}) should be one octave lower than the centre key on the lower string ($S_{g'}$).

Those who have access to a frequency counter may benefit from Table 2. This lists the correct frequencies for all keys, with the instrument set to the highest octave (S2 and S3 closed, S4 open). As before, the presets in each chain must be adjusted from 'high' to 'low': from top to bottom in the Table, in other words.

Play

Those who can play the guitar should have no difficulty 'picking up' this instrument: it is meant for them! As an

aid to the less-experienced, figure 11 lists 24 useful chords. Just as a starter: there are many more — but most people work out their own preferences.

Instructions for use: the keys that are marked with a dot are depressed, and then all five string keys are operated. The 'open notes' are also required, so S1 should be set to position a. ◄
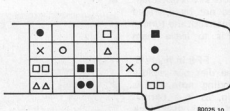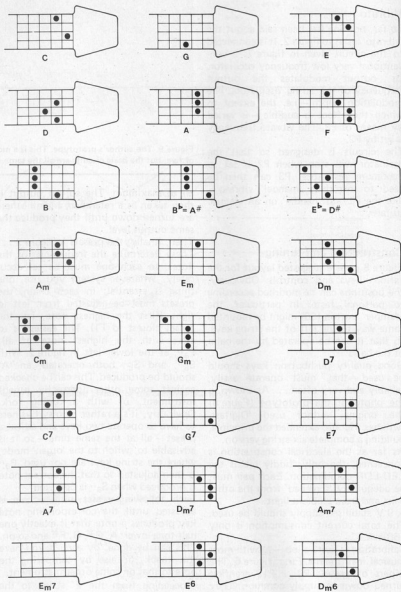
10

Figure 10. As an aid to tuning, this drawing illustrates which keys should produce the same note. Two keys with the same symbol(s) should sound identical; if one key has a certain symbol and another has two, this indicates that the latter key should sound exactly one octave higher.

11

Figure 11. These twenty-four basic chords should prove a useful aid for beginners.

There's no accounting for taste, so they say. The Elektor piano seems to be an exception: several readers have called us to account for the fact that it didn't sound the way it should. Perhaps this was only to be expected: we were quite satisfied when it sounded like other electronic pianos — but most people want it to sound like the real thing.

We have listened to our readers' comments; we have listened to the piano — modified it — and listened to it again. To be honest, we tried several modifications and rejected most of them because they were either too extensive

R4, R5, R10, R11, R16, R17, R22, R23, R28 and R29 are deleted.

Modifying the boards is a fairly simple matter, especially since all changes are clearly shown in figures 3 and 4. First, the octave board (figure 3). P1 and R37 can be removed (or left in, if you like — it doesn't make any difference) and a wire link is added between the wiper connection of P1 and the right-hand end of R37. This connects the original output rail (and the original output connection) to supply common. *The output is now taken from the −U1 connection at the upper right-hand edge in the component layout in figure 3.*

a piano that sounds like a piano

a minor change for a major improvement

Good news for those who have built the Elektor piano! A few simple changes and one or two additional components make it sound much more 'realistic'. Less like an electronic piano and more like the real thing.

or just not worth the effort. Now, finally, we've found what we were looking for: a few simple changes that make all the difference.

Operation: piano

The proposed modifications affect both the octave boards (figure 7 in the original article, modified as shown in figure 1 here) and the filter circuit (figure 13 in the original, figure 2 here). On comparing the new version of the keyboard circuit with the original, the differences will be obvious. The original output (the common rail linking R25 to R36) is now connected to supply common; this makes P1 and R37 redundant, so they can be omitted. Obviously, it is a good idea to have an output somewhere. This is achieved by adding a preset potentiometer between the commoned collectors of T1 . . . T12 and the −13 V rail (U1). Effectively, in other words, the output is moved from one end of the resistor/electronic switch/transistor groups to the other. That's all for the octave circuits.

In each of the five input lines to the filter circuit (figure 2), electrolytic capacitors are added (C47 . . . C51). Furthermore, four capacitor values are changed: C1 now becomes 6n8, C2 becomes 27 n, C3 is now 47 n and C4 is also changed to 47 n. At the outputs of opamps A1 . . . A5, several components are removed: diodes D1 . . . D5 are replaced by wire links, and resistors

The left-hand '−U1' connection retains its original function. For this modification, the copper track beside pin 7 of IC3 must be cut. The easiest way is to make two parallel cuts in the track, using a sharp knife, the track between these cuts is then heated with a soldering iron, after which it can be lifted off the board with the tip of the knife. Furthermore, the wire link immediately to the right of IC1 and that to the left of IC2 are removed. The IC1 wire link is replaced by preset potentiometer P9; note that the wiper is connected to one end. Finally, the −U1 connection to IC2 and IC3 must be restored. Using insulated wire, the −U1 connection is linked to pin 7 of IC2 and pin 7 of IC3, as shown. Note that the −U1 connection to the next board may also have to be rewired — if it was taken from the upper right-hand edge of this board.

The filter board is even easier. Diodes D1 . . . D5 are replaced (or bridged) by wire links. Resistors R4, R5, R10, R11, R16, R17, R22, R23, R28 and R29 are removed. Capacitors C1 . . . C4 are replaced, using the new values given above. Finally, electrolytics are included in series with the inputs — the neatest way to do this is by breaking the copper tracks near the inputs, as shown, and mounting the capacitors across the gap. That just about wraps it up. The only remaining point is the adjustment procedure for the five new potentiometers (P9). Initially, they are set to minimum resistance; then they are turned up to the point where even
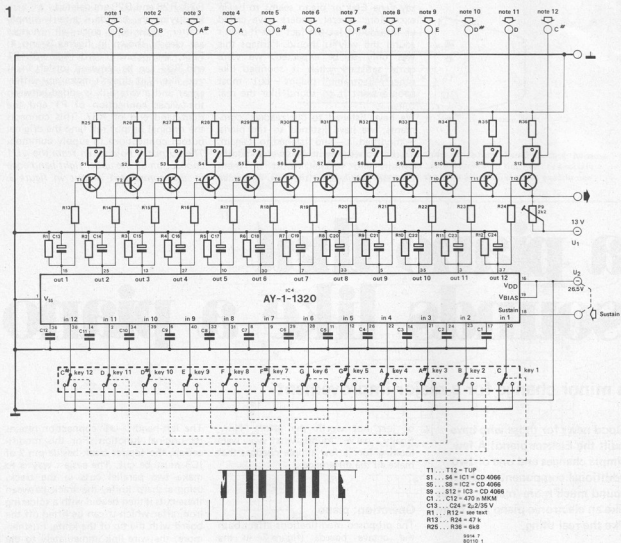
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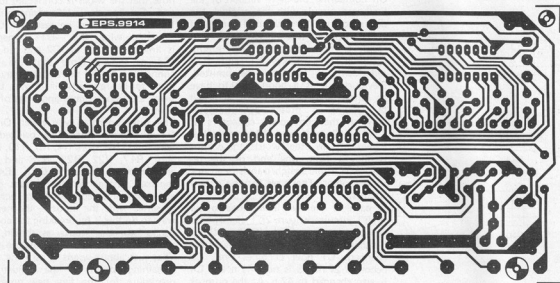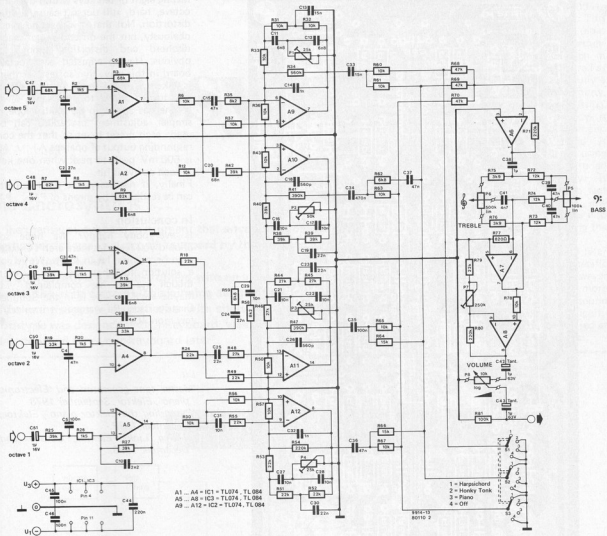Figure 1. Modified 'octave' circuit.

T1 . . . T12 = TUP
S1 . . . S4 = IC1 = CD 4066
S5 . . . S8 = IC2 = CD 4066
S9 . . . S12 = IC3 = CD 4066
C1 . . . C12 = 470 n MKM
C13 . . . C24 = 2,2/35 V
R1 . . . R12 = see text
R13 . . . R24 = 47 k
R25 . . . R36 = 6x8

3

Figure 3. Modified 'octave' p.c. board.

2

Figure 2. Modified filter circuit.

A1 ... A4 = IC1 = TL 074, TL 084
A5 ... A8 = IC3 = TL 074, TL 084
A8 ... A12 = IC2 = TL 074, TL 084

octave 5
octave 4
octave 3
octave 2
octave 1

TREBLE

BASS

VOLUME

1 = Harpsichord
2 = Honky Tonk
3 = Piano
4 = Off

4
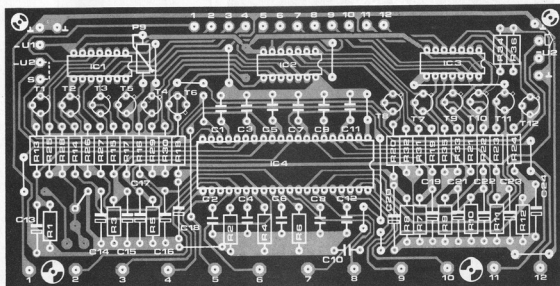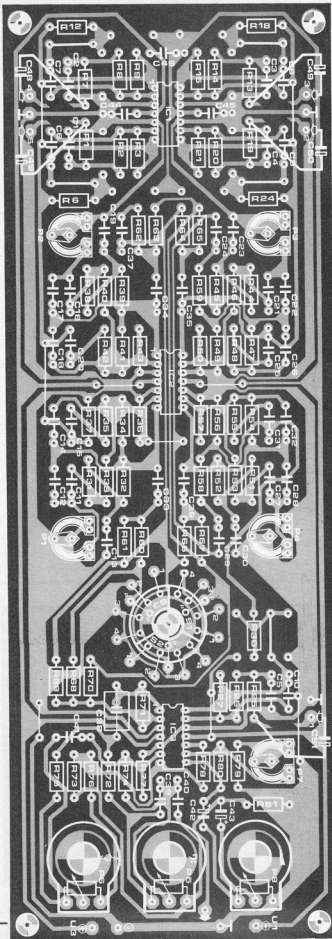
hitting eight or ten keys within the same octave, hard, still doesn't cause audible distortion. Not that it will sound 'nice', obviously, but the difference between a dischord and distortion should be obvious. Having adjusted each octave board in this way, the 'softest' of them is taken as a reference, and all others are turned back to the same level. If one has access to an oscilloscope, a simpler adjustment procedure can be used: each preset is set so that the corresponding output of opamps A1 . . . A5 is 500 mV peak to peak when one key in each octave is hit.

Finally, if necessary, the filter circuits can be readjusted by means of P1 . . . P4.

In conclusion

Is everybody happy? We hope so. Should you feel that there aren't enough even harmonics, let us know. We've got a solution for that one, too — even though no-one has complained yet. Maybe we'll include it in the Summer Circuits issue. Be warned: it involves a few more ICs.

Lit.

'Master tone generator' and 'Electronic piano', Elektor, September 1978.
'Extending the Elektor piano', Elektor, December 1978.
'Missing Link', Elektor, October and December 1978.

Figure 4. Filter board modifications.

the chorosynth

J.D. Mitchell

the Chorosynth is . . .

an inexpensive keyboard instrument that anyone can play, a mini synthesiser for use as a second keyboard by the working musician, or anything in between.

The Chorosynth is a real instrument with an amazing variety of sound possibilities at its disposal. The winning design of our competition, it is, as the name suggests, a synthesiser with chorus effect. Although the prototype was designed with a keyboard, we have made it possible to 'play now and buy your keyboard later'.

The dynamic range of the instrument needs to be between 150 and 3000 Hz at least, before it can be played. This corresponds to the length of organ pipes ranging from 16' to 2' (feet). Furthermore, the bass notes must be rich in harmonic tones which can be filtered as required. Now a tune played on only a few notes, however rich in harmony, sounds as though something is missing. However, when the same melody is played with fifths a much richer and therefore more musical sound is produced. If the tune is played using several sound sources in unison (tuned to virtually the same frequency), it sounds quite different and this in fact produces the chorus effect. A low frequency modulation (vibrato) is also particularly useful when simulating the sound of stringed instruments.

What does the Chorosynth sound like? As always the description of any specific sound is extremely difficult, however we are sure that the reader will be more than satisfied with the overall sound quality of the prototype. It is significant that the prototype has been used at live performances with great success.
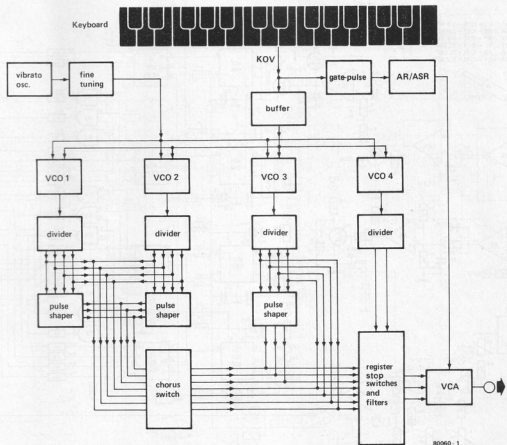
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Figure 1. The block diagram of the Chorosynth.
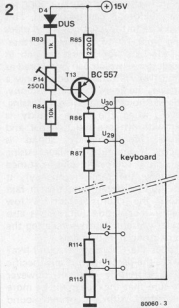
Figure 2. The current source and voltage divider chain for the keyboard of the Chorosynth.

The Chorosynth uses a 2½ octave keyboard and has a dynamic range of 6 octaves. Because of this broad frequency range, instruments ranging from violin to cello and flute to clarinet can be imitated. It is also possible to adapt the dynamics of the output signal with an envelope generator that can be switched to AR (attack-release) or ASR (attack-sustain-release) envelopes.

Block diagram

The block diagram of the Chorosynth is shown in figure 1. The 2½ octave keyboard produces a KOV (keyboard output voltage) which controls four VCOs (voltage controlled oscillators). VCOs 1 . . . 3 have the same voltage to frequency characteristics and are tuned to the same pitch. The fourth oscillator (VCO 4) is tuned to a frequency 1½ times higher, in other words it produces a note which is a fifth higher than that of the other VCOs. A fifth oscillator (marked vibrato in the block diagram) produces a relatively low frequency signal which is fed to the modulation inputs of the VCOs. The VCO output signals will then be slightly frequency

modulated to produce a vibrato effect. The frequency as well as the modulation depth can be varied.

The output of the VCOs are divided by 2, 4, 8 and 16. The dividers' outputs are exactly one octave apart and can be selected with the aid of 'stops' (or switches if you are not into organs!).

The output signals of the dividers and pulse shapers of VCOs 1 and 2 are added together and mixed with those of VCO 3 via the chorus switch. In this way the chorus effect can be switched on or off as required. The pulse shapers are simply a set of NAND gates which produce a square-wave with a duty-cycle of 25% from the divider outputs. It is this type of waveform that is particularly suitable for simulating the sound of stringed instruments. Of course, some filtering of the signal is required before it is acceptable to the human ear. The Chorosynth has two filters, one for strings and one for woodwinds.

A gate-pulse is derived from the KOV which is used to trigger the envelope generator. This can provide one of two envelope patterns — attack-release or attack-sustain-release. This envelope controls the VCA (voltage controlled
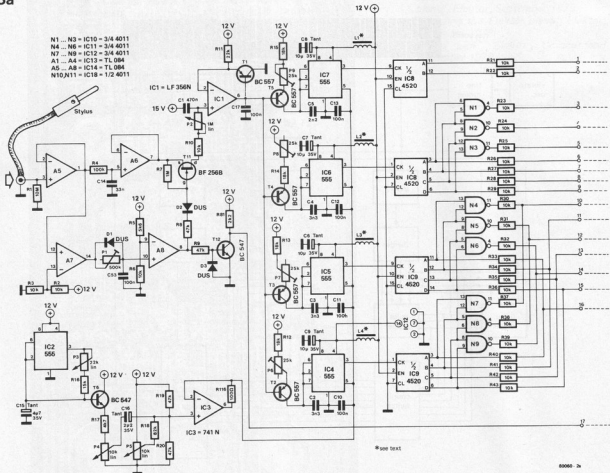
Figures 3a and 3b. The main circuit diagram of the Chorosynth, drawn in two sections for clarity.

amplifier) before being fed to an external amplifier.

Circuit diagram

The current source and voltage divider chain for the keyboard (shown in figure 2) provide each key with a specific voltage level. This keyboard output voltage (KOV) is then passed to the non inverting input of op-amp A5 in the main circuit diagram (figures 3a and 3b). This can be done by means of the 'printed circuit keyboard' (figure 4) and a stylus; alternatively, a conventional keyboard can be used.

The FET T11 functions as a switch to 'sample-and-hold' the keyboard voltage (in C1). The rate of change of the voltage level at the non-inverting input of IC1 is controlled by P2 and C1 allowing a glissando effect between notes.

The four VCOs in the block diagram are the circuits around IC4 to IC7. They are 555 timers which have been wired as astable multivibrators. ICs 4, 5 and 6 (VCOs 1, 2 and 3 in figure 1) are tuned to the same frequency and therefore have identical component values. IC7

3c
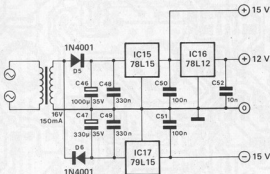
Figure 3c. The regulated power supply for the Chorosynth is included on the printed circuit board.

3b

4
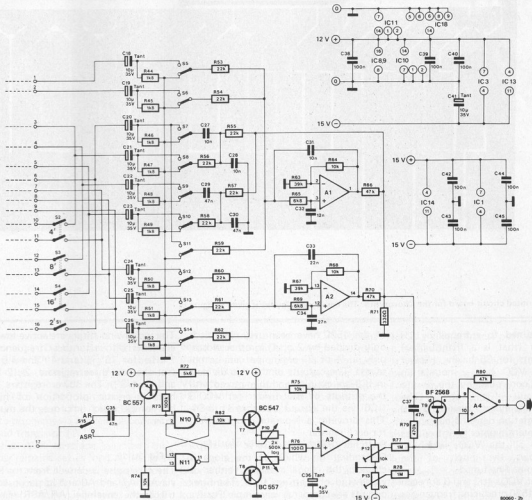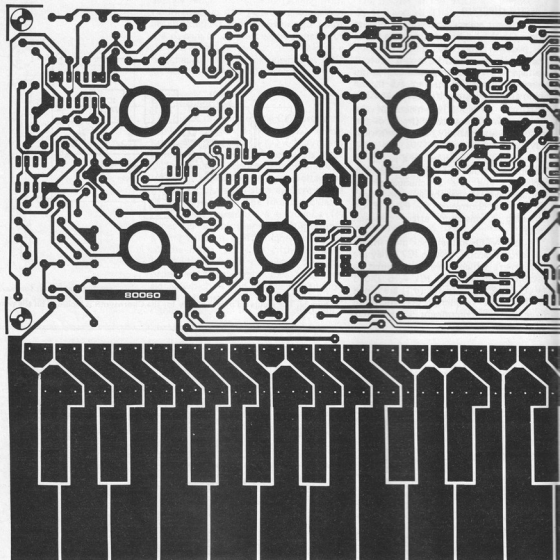
Figure 4. The printed circuit board for the Chorosynth. The component layout is given in figure 5.

(VCO 4) is tuned to a frequency 1½ times greater (that is a fifth higher) which accounts for C5 having a lower value. Each VCO has a modulation input (pin 5) controlled by the vibrato oscillator circuit — ICs 2 and 3. The frequency of the vibrato is varied by P3 and the modulation depth is controlled by P4. The potentiometer P5 introduces an offset to all the VCOs which in practice allows the pitch of the Chorosynth to be fine tuned.

The output of VCOs 1, 2 and 3 are each divided by a 4520 producing frequencies which are 2, 4, 8 and 16 times lower than the VCO frequencies. As far as VCO 4 is concerned, only two outputs

of the 4520 have been used. Thus it is only divided by 2 or 4; in other words, only a 1' or 2' organpipe can be obtained. The outputs of the two dividers in IC9 are coupled and then mixed with the outputs of the divider for VCO 3 (IC6) via the 'chorus' switches S1 to S4. This provides a seperate chorus effect for each register.

Switches S5 to S14 serve as stops. Depending on which of the stops is closed, the signal reaches either a woodwind or a strings filter. The filter circuits are comparatively simple. Passive highpass filters whose top-end response is slightly rolled off by capacitors C28 and C30 provide the voicing for strings.

The woodwind filters are active lowpass elements with a turnover frequency of 2 kHz (for 16', 8' and 4') and 4.5 kHz (for the top three registers, 2', 2 2/3', and 1 1/3'). The lower registers thus have a greater proportion of higher harmonics, which improves the musical tone.

Gate pulse

The gate pulse is derived from the KOV via A5, A7 and A8 and its purpose is to trigger the envelope (AR/ASR) circuit. The type of envelope contour is selected by switch S15. With this switch in the AR position, the positive going edge of
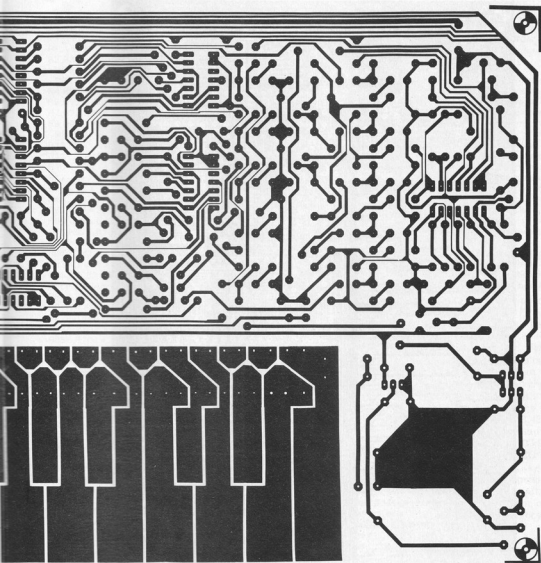
the gate signal triggers the flip-flop formed by NAND gates N10/N11, turning on T7 and charging capacitor C36 via the attack control, P10. As soon as the voltage on C36 reaches approximately 13.5 V, T10 turns off and the flip-flop is reset. The capacitor then starts to discharge via the release control, P11, and transistor T8.

With the ASR envelope selected, the flip-flop remains set as long as the gate signal is present, that is as long as a note is held (sustained) on the keyboard. Only when the key is released can T10 reset the flip-flop and C36 discharge (release).

The output of the envelope shaper circuit controls a simple VCA (voltage controlled amplifier), which in turn determines the dynamic amplitude characteristics of the output signal. The VCA consists of an op-amp (A4), with a FET (voltage controlled resistor) connected in the feedback loop.

Construction and setting up

The printed circuit board for the Chorosynth (figure 4) includes all the components and a keyboard layout. The keyboard has been included for those readers who wish to keep the cost down to a minimum. It is a simple matter to connect a conventional keyboard, one

set of make contacts per key is all that is required.

It should be noted that since the oscillators have a linear voltage-frequency characteristic, the keyboard tuning resistors must form a logarithmic potential divider. The appropriate values (for example R86a and R86b) are all made up using resistors from the E24 series. With 1% tolerance resistors, a tuning accuracy of 1% of a semitone is obtained, however (with the exception of R115a and R115b where 1% must be used) 5% resistors will also prove suitable, since the chorus effect by and large obscures any slight mistuning. Inductors L1 to L4 are ferrite beads

5
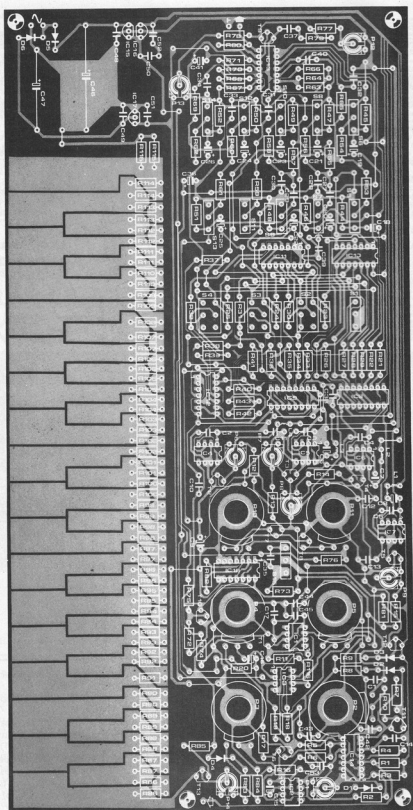
Figure 5. Component layout. In order to fit it on this page, this layout is reduced in size.

Parts list

Resistors:
R1 = 10 M
R2,R3,R6,R10,R21 . . . R43,
R64,R68,R74,R75,R82,
R84 = 10 k
R4,R73 = 100 k
R5,R72 = 5k6
R7,R77,R78 = 1 M
R8,R9,R19,R20,R66,R70,
R80 = 47 k
R11,R53 . . . R62 = 22 k
R12 . . . R15 = 18 k
R16 = 15 k
R17 = 4k7
R18 = 82 k
R44 . . . R52 = 1k8
R63,R67 = 39 k
R65,R69 = 6k8
R71,R978,R112b = 330 Ω
R76 = 100 Ω
R81 = 2k2
R83 = 1 k
R85,R105B = 220 Ω
R86A,R90A = 6Ω2
R86B,R101B = 560 Ω
R87A = 6Ω8
R87B = 150 Ω
R88A,R89A = 10 Ω
R88B,R108 = 22 Ω
R89B,R110A,R111A,
R113A = 27 Ω
R90B,R948,R106B = 11Ω5
R91,R94A,R109B = 8Ω2
R92A = 9Ω1
R92B,R103B = 180 Ω
R93A,R96A,R97A,R99A,
R100A = 12 Ω
R93B = 39 Ω
R95A,R98A,R101A,
R109A = 15 Ω
R95B,R114A = 33 Ω
R96B = 120 Ω
R98B = 68 Ω
R99B = 1 Ω
R100B = 1Ω8
R102A = 16 Ω
R102B,R104B,R114B = 470 Ω
R103A,R104A,R106A,
R107A = 18 Ω
R105A,R112A = 20 Ω
R107B = 27Ω7
R110B = 270 Ω
R111B = 680 Ω
R113B = 21Ω2
R115B = 120 k
R115A = 1k2 1%

Potentiometers:
P1 = 500 k (470 k) preset
P2 = 1 M lin
P3 = 22 k lin
P4,P5 = 10 k lin
P6 . . . P9 = 25 k presets
P10,P11 = 500 k (470 k) log

Capacitors:
C1 = 470 p
C2,D3,C4 = 3n3
C5 = 2n2
C6 . . . C9,
C18 . . . C26 = 10 μ/35 V
tantalum
C10 . . . C13,C17,C38 . . . C45,
C50,C51,C53 = 100 n
C14 = 33 n
C15,C36 = 4μ7/35 V tantalum
C16 = 2μ2/35 V tantalum
C27,C28,C31,C52 = 10 n
C29,C30,C35,C37 = 47 n
C32 = 12 n
C33 = 22 n
C34 = 27 n
C46 = 1000 μ/35 V
C47 = 330 μ/35 V
C48,C49 = 330 n

Semiconductors:
D1,D2,D3,D4 = DUS
D5,D6 = 1N4001
T1 . . . T5,T8,T10,
T13 = BC 177B, BC 557
T6,T7,T12 = BC 107B, BC 547
T9,T11 = 2N3819, BF 256B
IC1 = CA 3140, LF 356
IC2,IC4 . . . IC7 = NE 555
IC3 = 741
IC8,IC9 = CD 4520, MC 14520
IC10 . . . IC12,IC18 = CD 4011
IC13,IC14 = TL 084
IC15 = 78L15
IC16 = 78L12
IC17 = 79L15

Miscellaneous:
Tr1 = 16 V/150 mA transformer
S1,S5 . . . S15 = SPDT
S2 . . . S4 = DPDT
L1 . . . L4 = 5 turns of 0.2 mm
dia. enamelled copper wire on a
ferrite bead.

Technical details:

Tonal		
Range:	C to c⁵	65.41 Hz to 4184 Hz

Keyboard:	2½ octaves	
Registers:	Cello (16')	S10
	Bassoon (16')	S14
	Viola (8')	S9
	Clarinet (8')	S13
	Violin (4')	S8
	Clarinet (4')	S12
	Violina (2')	S7
	Flute (2')	S11
	Twelfth (2 2/3')	S6
	Larigot (1 1/3')	S5

Effects:	Chorus (16')	S4
	Chorus (8')	S3
	Chorus (4')	S2
	Chorus (2')	S1

Additional controls:

Glissando: (Portamento)	P2
Vibrato: Depth and rate	P8 and P9
Envelope	
shaper:	Attack-release or attack-sustain release selectable by S15; attack and release times independently variable between 1 mS and 10 S (P10 and P11)

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with five turns of 0.2 mm enamelled copper wire. All other components are readily obtainable.

Since the Chorosynth has a large number of potentiometers, controlling it may at first be a little difficult. It is advisable to set all the controls to the following positions at the beginning of the setting up procedure:

P1	minimum resistance, fully clockwise.
P2	minimum resistance, fully anti-clockwise.
P3	not important
P4	wiper to earth, fully anti-clockwise.
P5	mid-position
P6	minimum resistance, fully anti-clockwise.
P7 . . . P10	mid-position
P11	minimum resistance, fully anti-clockwise.
P12 . . . P13	wiper to earth, fully clockwise.
S1 . . . S4	open
S5, S6	switched to ground
S7	switched to C20
S8 . . . S14	switched to ground
S15	ASR position

The VCA has two adjustment points, P12 and P13. P13 determines the minimum gain and is adjusted such that no output signal is audible. A note is then 'struck' and held, whilst P12 is adjusted until a slowly rising signal appears at the output of A4. P1 is then adjusted so that no change in frequency occurs when a key is struck.
In order to trim the keyboard, P14 is adjusted to give 8.43 V across R115. This should be measured with a universal meter of at least 10 kΩ/V. With the aid of P8, the third VCO (IC6) is adjusted, so that when the key to the far left of the keyboard is operated, a C2 note can be heard at the output. The pitch can be compared with that of another instrument, or the frequency can be measured (523.2 Hz).

S2 is then closed and, by using P7, the frequency of IC5 is trimmed to a value which should correspond as much as possible with that of IC6 — minimum beat note in the output signal. At the same time P6 must be trimmed so that there is very little vibrato at the output. It should now sound like a chorus. Finally, P9 is adjusted so that the output frequency of IC7 (VCO 4) is 1½ times that of the other VCOs. The Chorosynth should now be ready to play. Under 'technical details' the stops and their values are indicated.

Although this may be the end of the Chorosynth article it certainly is not the end of the possibilities for this instrument. It will be apparent to many readers that the Chorosynth offers considerable scope for many modifications. If you find one that you would like to share, we would like to know about it.

dual slide fader

for smooth, continuous projection

Enthusiastic amateur photographers and professionals are not only interested in making good pictures: they also want to present them properly. For a slide show, the professional 'trick' of using two projectors alternately is becoming increasingly popular — the 'black' gap between successive slides can be eliminated in this way. For the finishing touch, each following slide should be gradually blended into the existing picture. This is where the dual slide fader comes in: as the brightness of one projector is smoothly increased, the other is gradually faded out.

A complete dual projection installation consists of two slide projectors and a control unit. Complete, integrated units also exist — everything built into one case — but it is more common to use a separate control unit with two standard projectors. Most commercially available units have one disadvantage in common: they are expensive. The reason for this is that they are usually designed for maximum flexibility, with all kinds of additional 'features': automatic fading, tape-slide synchronisation, so-called 'sparkling' effects and so on.

However, if one reduces the number of applications and features, aiming at a manually operated dual fader, it is possible to come up with a simple and cheap design. The 'fade' from one projector to the other is done manually, by means of a stereo potentiometer; slide changing is also done by operating the two projectors in the normal manner, it is not done automatically by the fader unit.

Figure 1 gives a drastically simplified block diagram. It should be noted that there are two types of projection lamp: one operates at the full mains voltage, whereas the other (more modern) type uses a much lower voltage (24 V). In this article, two circuits will be described

for this reason: however, the block diagram is equally valid for both.

Mains voltage control

The circuit given in figure 2 works in the same way as a normal triac mains dimmer. For older projectors, using a mains-voltage lamp, this circuit is ideal. It can also be used for more modern projectors that use a 24 V lamp, varying the primary voltage to the lamp transformer in the projector. This is not an ideal solution, however, since a mains transformer is not designed to run on the type of distorted waveform that a triac control unit produces. The ratio between primary and secondary voltage is not constant over the full control range, making the control characteristic rather less smooth than one would like.

To sum it up: if you have older projectors, using mains-voltage lamps, read on; if you have modern projectors with 24 V lamps, skip this section . . .

The circuit itself is quite straightforward. Two standard triac control circuits are used, with the control potentiometers combined into one stereo potentiometer (P2). Note that a tandem 470 k *linear* potentiometer should be used. Both

control circuits can handle any load up to 400 W.

One control circuit can be mounted on the printed circuit board given in figure 3. For a complete unit, two of these boards are therefore required. The interference suppression coils (L1 and L2) must be able to handle a current of up to 2 A; this type of coil is available ready-wound. The two boards, two coils and the stereo potentiometer can all be mounted in a suitable plastic case.

Note that the whole circuit is connected to the mains! This means that all due care must be taken when constructing the unit. In particular, an insulating case must be used and the potentiometer should have a plastic spindle. The capacitors must also be suitable for the relatively high voltages present in the circuit.
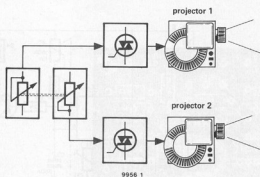
Figure 1. Simplified block diagram of a dual slide fader unit. The lamps in the two projectors are controlled by triac 'dimmers'; a simple (stereo) potentiometer is used to control both units.

Parts list for figure 3

Resistors:
R1 (R4) = 560 k
R2 (R5) = 2k7
R3 (R6) = 470 Ω
P1 (P3) = 100 k preset
P2 = 470 k linear tandem
potentiometer

Capacitors:
C1 (C4) = 22 n/400 V
C2 (C5) = 100 n/400 V
C3 (C6) = 100 n/600 V

Semiconductors:
Di1 (Di2) = 45412, ER 900
Tri1 (Tri2) = 41015, TIC 226

Sundries:
L1 (L2) = 100 . . . 200 μH/2 A
interference suppression coil
Note: two p.c. boards are required; the components given in brackets are mounted on the second board.
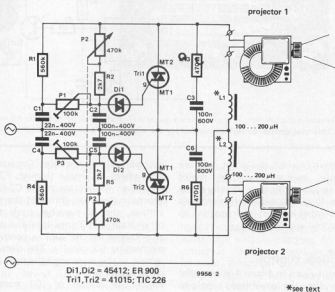
Figure 2. A mains-voltage control circuit. This is intended for use in projectors with lamps that run off the full mains voltage.
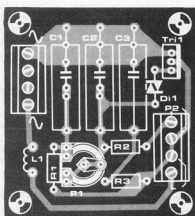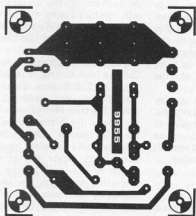
Figure 3. Half of the circuit given in figure 2 can be mounted on this board; two boards are therefore required for the complete unit. This means that it is possible to mount one board in each projector. However, great care should always be taken: the complete circuit — including the control potentiometer — is connected to the mains!

4
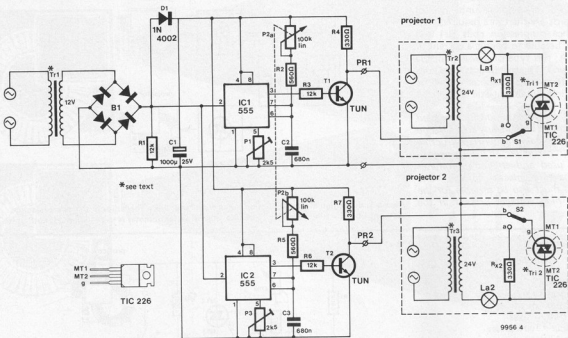
Figure 4. This triac control circuit is intended for use with low-voltage (24 V) lamps.

If you have reached this point, you are apparently interested in the mains voltage version. You may therefore pass over the following section (which deals with the other circuit) and proceed to 'construction and use'!

Low voltage control

The circuit given in figure 4 is intended for controlling the low-voltage supply to a 24 V lamp. As before, the two triacs are mounted inside the slide projectors and the two coupled control circuits are mounted on a single printed circuit board in the control box.

As before, the circuit consists of two complete triac control circuits, linked by means of a stereo potentiometer. The circuits are rather more sophisticated than those used in figure 2, since a much lower voltage must be regulated smoothly. Since both control circuits in figure 4 are identical, we will confine the explanation to the upper half.

Trigger pulses for the triac Tri 1 are produced by the well-known timer IC, type 555. In this circuit, it is used as a monostable multivibrator; it is triggered each time the voltage at pin 2 decreases to below approximately one-third of the supply voltage. Since this voltage is actually the full-wave rectified output from a bridge rectifier (B1), the IC will be triggered once during each half-cycle of the mains waveform.

When IC1 is triggered, its output (pin 3) goes 'high' – to the full supply voltage – and pin 7 (the 'discharge output')

goes basically open-circuit. Capacitor C2 now starts to charge through P2a and R2. When the voltage across C2 reaches approximately two-thirds of the supply voltage, the IC switches back to the other state: pin 3 goes 'low' and pin 7 is also connected to supply common – discharging C2 rapidly. The circuit will remain in this state until the next trigger pulse occurs at pin 2.

When the output of IC1 goes low, transistor T1 cuts off. The voltage at the collector of T1 swings positive, triggering the triac (provided S1 is in the position shown). The triac will now start to conduct, until the following zero-crossing of the mains waveform. Since the moment that the output of IC1 goes 'low' depends on the setting of P2a, this potentiometer effectively controls the point on the mains waveform where the triac starts to conduct. This, in turn, determines the brightness of the projection lamp, La1.

The other half of the same potentiometer (P2b) controls the brightness of the second projector. Provided a linear potentiometer is used, wired exactly as shown, the result is a smooth fade from one projector to the other.

Construction and use

As far as the construction of the unit is concerned, the main points for the mains voltage version have already been mentioned: adequate electrical insulation and sufficiently high voltage rating for the capacitors.

For the low voltage control unit, there are a few more points to note. The bulk of the circuit is mounted on the board given in figure 5; this, and the mains transformer (Tr1) can be mounted in a small case. A miniature 12 V transformer can be used, as the circuit only draws a few milliamps; however, switches S1 and S2 and resistors Rx1 and Rx2 are all mounted in the projectors; transformers Tr2 en Tr3 are the existing transformers in the projectors.

The switches are included so that the projectors can also be used on their own in the normal way. With S1, for instance, in position 'a' the corresponding triac will conduct during the full mains cycle – the projector will continue to work as if nothing had been modified. When the switch is in position 'b' the projector can be controlled by the slide fader unit. Quite a high current is required for the lamps (a 250 W lamp draws more than 10 A), and since some 1.2 V remains across the triac even when it is fully 'on' the power dissipation can be over 12 W. For this reason, each triac must be mounted on a fair-sized heatsink; if it is possible to locate this at some point where it is cooled by the fan in the projector, so much the better. When mounting the triacs, it should be noted that 'main terminal 2' (mt2, the cathode) is connected to the metal part of its case.

When it comes to actually wiring up either of the control units to a given pair of projectors, some electrical

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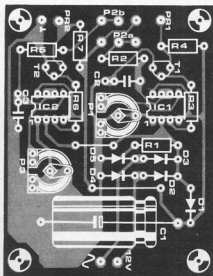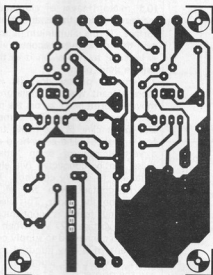
Figure 5. Printed circuit board and component layout for the circuit given in figure 4. The triacs, switches and resistors R_x are mounted in the projectors — using an adequate heatsink!

Parts list for figure 5

Resistors:
R1,R3,R6 = 12 k
R2,R5 = 560 Ω
R4,R7,R_{x1},R_{x2} = 330 Ω
P1,P3 = 2k7 preset potentiometer
P2 = 100 k linear tandem potentiometer

Capacitors:
C1 = 1000 μ/25 V
C2,C3 = 680 n

Semiconductors:
IC1,IC2 = 555
T1,T2 = TUN, BC 107
D1 = 1N4002
B1 = 4 x 1N4002
Tri1,Tri2 = TIC 226

Sundries:
Tri1 = 12 V/100 mA mains transformer
S1,S2 = single-pole switch

'peculiarities' are likely to be found. Since virtually every projector is different, it is impossible to discuss all possibilities in detail. However, the following points may be a help:
● Nearly all projectors contain a fan. In most cases, this is connected to the mains; sometimes it runs off the 110 V tap on the lamp transformer. It may also be found that the fan motor actually doubles as mains transformer!
● A remote-control slide-change mechanism is usually driven (via belt or gear wheels) by the fan motor, although more expensive projectors may have a separate motor for the job. When a slide is to be changed, a mechanical coupling is operated by an electromagnet; the latter is often powered from the 24 V winding on the lamp transformer. In some cases, a simple DC supply may be used for the magnet.
● Many projectors also incorporate some kind of 'back-one-step' slide change. This will also usually involve some electromagnetic mechanism that is powered from the low-voltage supply.
● Remote-control or automatic focussing ('autofocus') and/or timer mechanisms usually operate off a full-wave rectified 24 V supply. In some cases, an additional 12 V supply may be found for these features.
● In some slide projectors, it is possible to switch the lamp to a lower voltage — giving less light, but extending the lamp life considerably. Usually, a lower-voltage tap on the transformer is used for this.

It will be apparent from the points listed above that installing the mains voltage control unit, in particular, may call for quite extensive rewiring of the projector. If the lamp runs off the raw mains voltage, well and good — that's what this unit is designed for. If the projector uses a low-voltage lamp, however, things get complicated. Everything except the lamp must be disconnected from the transformer and powered from a separate low-voltage supply. After all, the idea is that slides are to be changed when the lamp is off — and at that time, the transformer is virtually disconnected from the mains! Installing the low-voltage control unit will rarely be a problem. In nearly all cases, at least one wire will be found that runs direct from the transformer to one side of the lamp (and to nothing else) — the other lamp lead may run through some switches, making it more difficult to trace. The triac can be connected in series with the direct connection to the lamp.

The matrix printer shown in the photo is the Matsushita EUY-1OE-1OT. This is a so-called electro-sensitive printer, that prints the characters (letters, numerals and symbols) on a special type of paper. This paper is coated with a very thin (0.1 micron) layer of aluminium; immediately below this is an almost black surface. If the aluminium is scratched off, this dark layer becomes visible; it is possible to 'write' on it in this way, using a sharp point.

The matrix printer does its job in a more elegant way. A miniature motor moves a 'print head' across the paper, as each line is printed. The same motor also takes care of the paper transport for 'line feed'. The print head contains a vertical row of seven little 'pins'; these can be connected (via electronic switches) to supply common. The pins are in contact with the aluminium coating on the paper. This electrically conducting layer is connected to a positive voltage (27 V). When one of the pins is connected to supply common, a current of a few hundred milliamps flows through the aluminium coating and the printer pin. At the point where the latter touches the paper, a high current density results. The heat produced is sufficient to burn away the aluminium layer at that point, leaving a small black dot. Since more than one of the pins can be connected to supply common simultaneously, it is possible to write up to seven dots in a vertical row at any time.

If voltage is now applied to the motor, the print head will move across the paper — writing up to seven horizontal lines, depending on the number of pins that are activated. At the end of the line, the printer head is automatically moved back to the beginning of the next line: 'line feed' and 'carriage return'.

printer for microprocessors

a black-and-white display

It can be useful to print the information output from a microprocessor system on paper. For small μP systems that give a hexadecimal output, a so-called matrix printer is a good choice. This is connected to the microcomputer system via a suitable interface.

In this article, a printer and interface for the SC/MP system are discussed, as well as the necessary software. A 'disassembler' program is also included — an invaluable aid when analysing existing programs. The printer and interface are suitable for other microprocessor systems as well; however, the software must then be modified, of course.

As the head moves across the paper, it is obviously possible to switch the pins on and off, producing dotted lines. In this way, all sorts of characters can be printed as groups of dots: the trick is to switch the pins on and off at exactly the right moments to produce the desired character. As shown in figure 1, each character is built up in a 35-dot matrix: five successive vertical columns (T1 . . . T5), each consisting of seven dot positions (N1 . . . N7, corresponding to the seven pins).

A character generator switches the printer pins on and off at the correct moments. This unit is part of the electronics that link the SC/MP system to the printer: the printer interface. This interface circuit will be discussed later.

The printer contains a reed switch to

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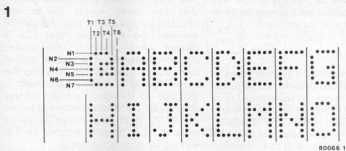
Figure 1. Using a 5 x 7 dot matrix, clearly legible characters are obtained.

indicate the end of a line, and the moment that the print head has returned to the beginning of the next line.

Furthermore, a 'sync generator' is included, intended for synchronising the character generator to the position of the printer head on the line. This 'generator' consists of a gear-wheel, driven by the motor in the printer and moving past the core of a coil. The core is magnetised, by means of a permanent magnet; as the teeth of the gear-wheel pass through the magnetic field, a sinusoidal voltage is included in the coil. The interface described here doesn't do the actual synchronisation itself; instead, the software is designed to ensure that the column spacing is maintained proportional to the speed of the printer head.

So much for the printer itself. On its own, however, without the interface circuit and corresponding software, it is useless as a typewriter without a typist. Which brings us to the next section:

The interface

A block diagram of the interface circuit is given in figure 2. The buffer memory (8-bit latch) is basically equivalent to a single RAM location. When the address decoder provides a 'store' pulse, the information present on the data bus is stored in this latch and passed (via suitable power drivers) to the printer
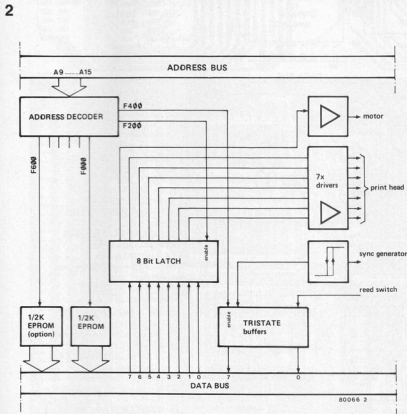
pins. Each 'store' pulse therefore enters the data for a single column of dots.

As mentioned earlier, the characters are printed one column at a time – five for each character. For each column, seven data bits are required ('stored') at an address in the field F200 . . . 3FFF; the eighth bit is used for motor control, via a separate buffer stage: it ensures that the motor starts at the correct moment at the beginning of a line, and stops immediately after returning to the beginning of the next line.

The address decoder is designed for use with ½ K address fields. The EPROM with the firmware for SC/MP system control and character generation is located between addresses F000 and F1FF. The following ½ K is used for addressing the 8-bit latch; then another ½ K (from F400 to F5FF) correspond to a tri-state buffer that outputs the data from the reed switch and sync generator in the printer to the data bus. The fourth address field (F600 . . . F7FF) can be used for an additional EPROM. This could be used, for instance, to store firmware that makes the printer routine independent of Elbug in the original SC/MP system – the push and pull routines, in particular. Alternatively, a special printer program could be located here. All in all, the complete interface uses a 2 K address field – half a page, in other words.

As mentioned earlier, the firmware takes care of character generation, including correct column spacing as determined by the speed of the printer head. Furthermore, it ensures that the text is printed in lines, each containing 8, 16 or 32 characters, as required. The SC/MP system must supply the data to be printed, one complete line at a time. To this end, the text must be loaded in ASCII code in a specified RAM area. Any available area can be used: as long as the interface is told where to look for it!

The character generator, stored as firmware, can generate 64 different characters. Only the lower 6 bits of the ASCII code in each RAM location are needed to select the correct character. The two remaining (upper) bits are used to define the desired character size, corresponding to 8, 16 or 24 characters per line. As an example, the letter A in 6-bit ASCII code corresponds to 01. When the two upper bits are both 0 (complete code: 01) the smallest character size will be printed, corresponding to 32 characters per line. However, if one of the two upper bits is '1', larger characters are obtained: '41' for an A corresponds to 16 characters per line, and '81' makes the A even bigger, for 8 characters per line.

When the firmware routine is started, it will want to know where to look in RAM for the text buffer that contains the data for one line of text. It will then scan the (hexadecimal) codes for the characters, as found in the text buffer, and convert them (by means of a

2

Figure 2. Block diagram of the printer interface.

'look-up table') from ASCII code to printer pin control data. The routine transfers the data for the five columns, in turn, to the latch; the character is printed.

Each character is automatically followed by a two-column space, after which the next character is retrieved from RAM and printed. This process is repeated until the end of the line (up to 32 characters). As each line is printed, the reed switch is 'open'; via N6, data bus line 00 is therefore 'high'. At the end of the line, the reed switch closes and the corresponding data line goes 'low'. This causes the program to load the text buffer, with the data for the next line; however, the printer head will not have got back to the beginning of the new line yet. The program continues to scan the data line, until the head is correctly positioned and the pins are again in contact with the paper. At that point, the reed switch opens and the next line can be printed.

Further details on the use of the text buffer and printer routine will be given when we come to the software.

The circuit

The complete circuit is given in figure 3. IC1 is the address decoder. It is connected to the upper seven address lines (making FFFF the highest address) and to the NWDS + NRDS line. Basically, this IC is a 3-to-8 decoder, but only half of it is used – as a 3-to-4 decoder. Output Y0 selects an EPROM, IC2, from address F000. This EPROM is also connected to the lower nine address lines, so that ½ K addresses are available. The output from the EPROM goes straight onto the data bus.

Outputs Y1 . . . Y3 of IC1 correspond to three further ½ K address fields: F200 to F400, F400 to F600 and F600 to F800. When output Y1 is active (and NWDS), the character information on the data bus is passed via IC6 to IC7.

The latter contains seven power drivers that can easily supply the current required for the printer pins. R8 and C6 are included to ensure that IC6 is cleared when power is first applied ('power-on reset'). The eighth output from IC6 is used to start and stop the motor, via T1 . . . T4. This circuit is designed to introduce a slight voltage drop, since the motor is intended to run on 24 V. When the motor is to be stopped, it is actually shorted by T4, making for a more rapid stop.

Output Y2 of IC1 enables the tri-state buffers, N5 and N6, when the third address field is selected (from F400). These buffers pass the information from the reed switch and sync coil in the printer to the data bus.

The power supply for the printer is a standard circuit, as shown in figure 4. The output voltage can be adjusted (by means of P1) in the range from 20 . . . 30 V. This gives some measure of 'copy darkness' control. A slightly
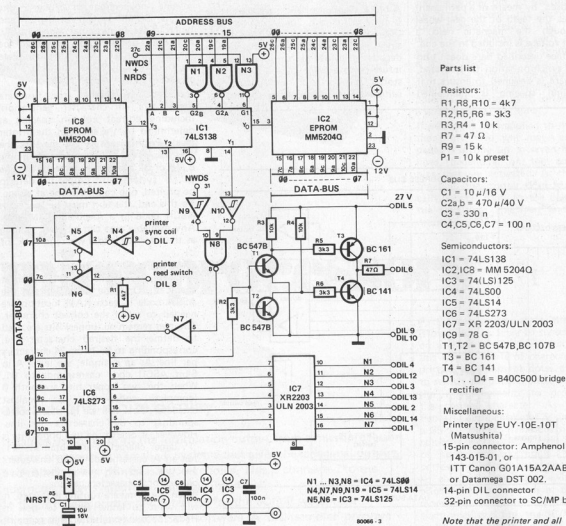
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Figure 3. The complete circuit. IC7 drives the printer pins, T3 and T4 control the motor.

Parts list

Resistors:
R1,R8,R10 = 4k7
R2,R5,R6 = 3k3
R3,R4 = 10 k
R7 = 47 Ω
R9 = 15 k
P1 = 10 k preset

Capacitors:
C1 = 10 μ/16 V
C2a,b = 470 μ/40 V
C3 = 330 n
C4,C5,C6,C7 = 100 n

Semiconductors:
IC1 = 74LS138
IC2,IC8 = MM 5204Q
IC3 = 74LS125
IC4 = 74LS00
IC5 = 74LS14
IC6 = 74LS273
IC7 = XR 2203/ULN 2003
IC9 = 78 G
T1,T2 = BC 547B,BC 107B
T3 = BC 161
T4 = BC 141
D1 . . . D4 = B40C500 bridge
rectifier

Miscellaneous:
Printer type EUY-10E-10T
(Matsushita)
15-pin connector: Amphenol type
143-015-01, or
ITT Canon G01A15A2AABL,
or Datamega DST 002.
14-pin DIL connector
32-pin connector to SC/MP bus

Note that the printer and all
connectors are available from
Technomatic Ltd.

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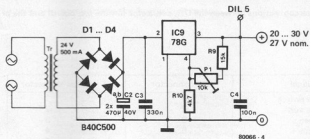
Figure 4. A simple stabilised power supply.

higher voltage than the nominal 24 V will do no harm to the printer head: those pins have 10 times the life expectancy of the rest of the printer mechanism! The motor is protected by means of series resistor R7 (figure 3), so that the power supply can safely be set to about 27 or 28 V. This has proved to give good results in practice.

Construction

The printed circuit board and component layout are given in figure 5. It is

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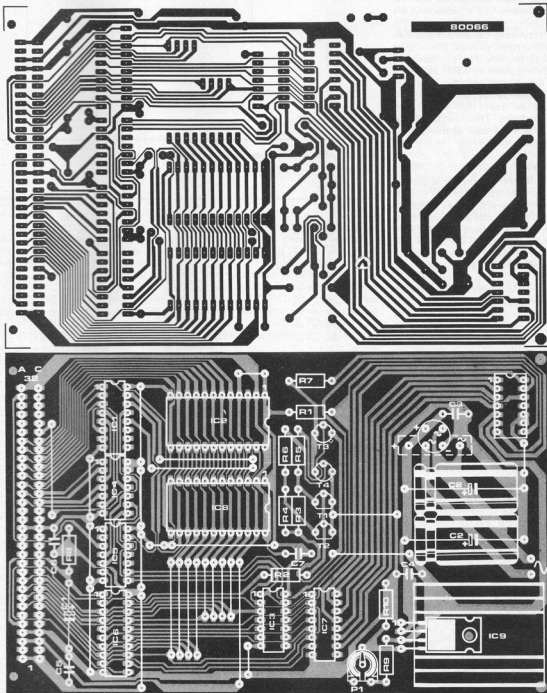
Figure 5. This p.c. board contains the complete interface and power supply. It plugs into the SC/MP bus.

Eurocard size, and fitted with a 64-pin connector so that it can be plugged straight into the main bus in the existing SC/MP system.

A 14-pin DIL connector at the other short edge of the board is used for the connections to the printer. The printer itself uses a 15-pin connector. The connections between the latter and the 14-pin connector on the p.c. board are listed in Table 1; for a neat appearance, it is a good idea to use so-called flat cable.

Since the power supply circuit is on the p.c. board, two connections for the mains transformer (24 V secondary) are also provided.

The complete unit can be plugged into the SC/MP system, without affecting its operation in any way. There is, however, one important point to note: *the metal case of the printer and the paper are both connected to the +27 V supply, so they must be kept well away from supply common*. That includes, for instance, the metal case of the existing SC/MP system . . .

The software

Without software, the printer is useless. The basic principle of the printer routine was mentioned earlier; a complete listing is given in Table 2. An extremely useful extension is given in Table 4: a disassembler program. An interesting point: both of these Tables were actually printed with the aid of these two routines!

The 'instructions for use' of this software are best broken down into a few groups.

Motor control

The latch in the interface can be addressed at F200. Storing data there (by means of the instruction MOD F200) with a value between 80 and FF starts the motor and causes up to seven horizontal lines to be printed.

The motor can be stopped by means of the NRST key, or by storing data between 00 and 7F in the latch.

Printer routine

As mentioned earlier, the text is printed one complete line at a time; all data for the line must be available in a text buffer (somewhere in RAM) before starting the printer. The location of this text buffer can be specified as required, provided the printer routine is started at address F200 (see Table 2). Alternatively, the routine can be started at address F00F; in this case, address 0F00 is automatically selected as the first address of the text buffer — in other words the ASCII character codes must be stored from 0F00 on, so that the interface can retrieve them from there as the line is being printed.

If the text buffer is to be located elsewhere in RAM, the first address must be specified in the TEXTAD locations: 0FF4 + 0FF5. The printer

Table 1. Interconnections between the DIL connector (on the p.c. board) and the printer.

signal	DIL connector	printer connector
reed switch	8	1
sync coil	7	2
screen	9	3
motor +	6	4
motor GND	10	5
paper contact (27 V)	5	6
NC		7
pin 7	11	14
pin 6	14	13
pin 5	2	12
pin 4	13	11
pin 3	3	10
pin 2	12	9
pin 1	4	8
NC	NC	15

Table 2. Listing of the printer routine, as stored in EPROM.

Table 3. This little program illustrates the 'instructions for use'.

```
                1 0
0000 08    NOP
0001 C4FF  LDI 0F      ; set P1 to
0003 35    XPAH 1      ; 0FF0
0004 C4F0  LDI F0
0006 31    XPAL 1
0007 C4FF  LDI 0FF     ; to 0FFF (PRINT-ROUTINE)
0009 C44C  LDI 0C
000B 37    XPAH 3
000C C4F0  LDI F0
000E 33    XPAL 3
000F C4F8  LDI F8      ; set TEXTAD (0FF4+0FF5)
0011 C9F4  ST -0C(2)
0013 C438  LDI 38      ; to 0838 (TEXT-TABLE)
0015 C9F5  ST -0B(2)
0017 C4FF  LDI 0FF
0019 37    XPAH 3      ; JS 3 (PUSH)
001A C455  LDI 55      ; set P3 to 0056-1
001C 33    XPAL 3      ; and JUMP via XPPC
001D 3F    XPPC 3
001E 00    HALT
001F 90FE  JMP $-1     ; JUMP
0021 00    HALT
0022 00    HALT
0023 00    HALT
0024 00    HALT
0025 00    HALT
0026 00    HALT
0027 00    HALT
0028 00    HALT
0029 00    HALT
002A 00    HALT
002B 00    HALT
002C 55    BYTE 55     ; SCRATCHPAD for
002D 55    BYTE 55     ; PRINTER-ROUTINE
002E 55    BYTE 55
002F 55    BYTE 55
0830 20 53 43 4F 50 2E 2E 20      ; TEXT-TABLE
0838 50 52 49 4E 54 45 52 2E
```

routine is then started at address F000. It should be noted that the printer routine uses a further two RAM bytes as scratch pad for counting; these bytes correspond to the five bytes immediately preceding the first address of the text buffer.

Short texts, less than one line in length, should be concluded by storing FF in the text buffer. This corresponds to a space, so that no further (unwanted) characters are printed. For that matter, it is good practice to conclude even complete lines with FF, as otherwise the printer may add part of a 33rd character at the end of the line.

After loading the text buffer, and the corresponding first address if necessary, the next step is to jump to the printer routine. This can only be done via the LIFO stack in the Elbug routine. This has the advantage that Elbug stores the CPU status in the stack before executing the printer routine. At the end of this routine, Elbug returns to the main program after restoring the original CPU status.*

The printer routine is started as follows. The start address of the printer routine minus one (i.e. F000 - 1 = FFFF — same page! — or F00F - 1 = F00E) is stored in ROUTAD (Routine address: 0FFC + 0FFD). Then a JS3 (PUSH) = 0056 - 1 instruction is given. An example is listed in Table 3; the

comments given there should give sufficient explanation. In this example, 0E30 . . . 0E40 is the text buffer; it contains the text '.SC/MP . . PRINTER.', where each '.' corresponds to a space.

If the same text buffer locations are used for the next line, the address specified in TEXTAD can be left. Obviously this data must be modified, however, if a different RAM area is to be used for the following line. Furthermore, it is possible to initiate further jumps to the printer routine by means of the instruction XPPC3, provided the contents of pointer register P3 are modified by the main program.

SC/MP disassembler

A disassembler routine can be used to analyse an existing program, and print a 'listing' by means of the printer routine. As mentioned, the 'raw' machine-code programs for the printer routine and disassembler routine were converted into Tables 3 and 4 in this way.

The disassembler first locates PC-relative jump instructions in the program to be analysed, calculates the initial addresses of the corresponding routines and marks these with a $ symbol and a number. This, in itself, breaks the program up into small sections — a great help when studying and trying to analyse an 'unknown' program. The complete program is printed as a series of addresses and mnemonics, as shown in the Tables. The disassembler routine starts at address 2600; the display then reads da-d-t-s. There are now three ways of executing the program: operating the D key (on the hex keyboard) initiates a print-out via the printer described above.

The T key (block transfer) is used for an output to the Elekterminal (1200 BD at flag 0). In this case, the halt LED will light every 16 lines; operating the halt/reset key initiates the print-out of the next 16 lines, and so on. Finally, operating the S key gives a serial output at 110 baud via flag 0; in this case, the print-out is continuous: it doesn't stop after each group of 16 lines.

In some cases, determined by the characteristics of the peripheral equipment, the print-out may appear on alternate lines — in other words, every other line is left unused. In this case, the data in address 2516 can be modified from 3F to 08.

After the desired key (D, T or S) has been operated, the display will indicate the type of output that has been selected. The next step is to enter the first and last address of the program that is to be 'disassembled'. If the first address is x000, the last address should not be xFFF (due to a peculiarity in the program . . .); it must be xFFE or less.

If there are tables in the program, key T (block transfer) is operated: 'ta' now appears in the display, and the first and last address of the table can be entered. More than one table (up to 15) can be specified in this way, provided they are entered in the order in which they occur in the program. A single-byte table can be specified; furthermore, two or more tables that occur in a consecutive series of addresses may be specified separately. All tables are printed in hex matrix format. It is even possible to print out a complete program in this way.

If a program is to be examined in which the addresses of the tables are unknown, the disassembler can still be used for an initial listing. Where tables occur, two question marks will be printed at the majority of the addresses, since the data will rarely correspond to a valid instruction.

Once the tables — if any — have been specified, the disassembler can be started by operating any key except T. It then scans the specified program (section) twice, locating the various subroutines on the first pass and printing during the second.

The disassembler can only handle programs that are contained on a single page. Larger programs must be dealt with one page at a time.

If certain locations are used as 'scratch-pad' memory, they will be indicated as 'byte' provided the data '55' is stored there before running the disassembler.

PC-relative jump instructions that specify an address outside the program range (lower than the start address, say) are not permitted.

The disassembler uses memory locations from 2840 up to 2A40 as scratch pad for 'lable' addresses: room enough for 255 PC-relative jump instructions.

A useful tip

The EPROM that contains the printer routine is not strictly necessary. The

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* Further details of the Elbug monitor routines are contained in Elektor, February 1979, and in the SC/MP book that will be appearing shortly.

Table 4. Listing of the disassembler program.

program given in Table 2 can be stored in RAM in the SC/MP system, say from address ØCØØ to address ØDFF. In this case, the instruction at address FØ5A must be modified accordingly: to C4ØC in this example, instead of C4FØ. In the demonstration program given in Table 3, the routine address will also have to be modified in this case: at address ØEØ7, data becomes C4ØB instead of C4FF. This is because the address of the printer routine minus 1 is stored at addresses 24EB and 24EF.

With these modifications, the three routines can be entered into the SC/MP and transferred from there to tape. If the printer routine is to be located elsewhere, the modifications will obviously be modified accordingly . . . Admittedly, this suggestion involves a lot of laborious loading-by-hand; but it saves an EPROM. It is the intention to include these routines on an ESS software record — at some future date. ◢

```
2465  ...
  (machine-code listing — multiple columns of hexadecimal
   addresses and assembly instructions)
```

remote control

Remote control units are becoming an almost standard accessory on colour TV sets. Hifi equipment that can be controlled 'remotely' is perhaps not commonplace — but it is no longer science fiction. Even modern slide projectors are going 'wireless'.

Obviously, tripping over wires is to become a thing of the past. Remote control systems — of the type described in this article — will become increasingly popular.
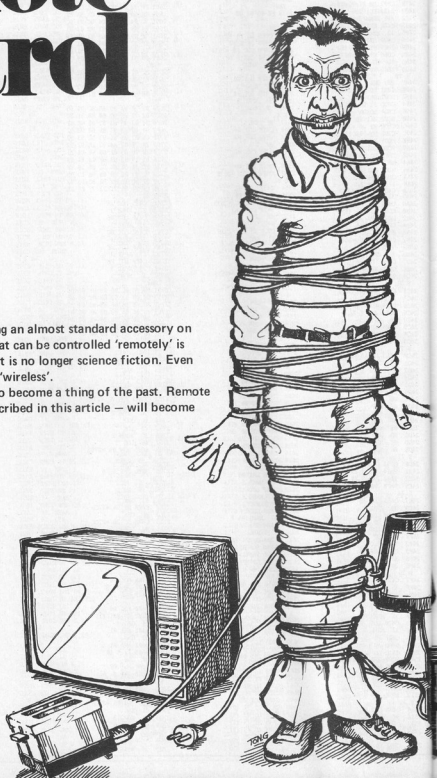
There may be all kinds of reasons for wanting to control some item of equipment from a distance. In domestic surroundings the most important reason, nine times out of ten, is a desire for increased 'ease'. There are also cases where remote control can be quite useful: adjusting the balance control on an audio system, for instance — the optimum setting can only be judged when you're in the listening position. From there, you can rarely reach the amplifier . . . There are even cases where some kind of remote control is essential. Model railway locomotives, for instance, will have to be controlled from a distance until somebody succeeds in manufacturing sufficiently miniaturised engine drivers.

Among remote control systems, the 'wireless' variety are the most useful. After all, there's not much fun involved in tripping over cables all the time — particularly when the other end of the cable is connected to some expensive item of equipment.

Whatever the reasons may be for wanting a wireless remote control system, this article gives some suggestions. The circuits are all intended for indoor use: they use an ultrasonic link, so that the range is too limited for model boats and aeroplanes. Remote control of television sets, hifi equipment, doors, lights, slide projectors — that is the kind of application where these systems can prove useful. A further application is in combination with the single-button control unit described in Elektor, June 1979: the Monoselektor.

Two systems

We will describe two different systems, both using an ultrasonic link. The first system was designed primarily for transmitting sixteen independent 'on/off type' commands. These command signals may be used to control all kinds of things: channel selection in a TV set, switching a lamp on or off, operating an electric door-opener, and so on. The sixteen command signals may all be received and decoded by a single receiver, but several receivers may also be used. For instance, it might prove useful to have one receiver for command signals 1 . . . 3, another for signal 4, a third receiver for signals 5 and 6, and so on. It is also possible to 'tune' several receivers to the same command signal. In the example given above, a fourth receiver might be added that will respond to commands 1, 4 and 6. It is not a good idea, however, to use more than one transmitter in this system — it would almost certainly cause confusion at the receiver end.

The electronics required for this sixteen-channel system is all of the 'available-off-the-shelf' variety: CMOS ICs, 555 timers, etc.

Not so for the other system: it uses two separate Plessey ICs, one for the transmitter and one for the receiver. This simplifies the circuits drastically. The ICs are intended for remote control of colour TV receivers, but they can prove equally useful in other applications. In principle, this system can transmit 32 'on/off' signals; however, some of these may also be combined to

The first system

First, let's discuss the system that uses standard components. After all, it was designed with the home constructor in mind!

Digital signals are easier to transmit reliably. 'Digital' means that they can represent only two conditions: on or off, yes or no, 'nought' or 'one'. These digital commands must be transmitted to the (remote) receiver; as with radio, this is done by modulating a high-frequency carrier signal. Two likely possibilities are illustrated in figure 1: Phase Shift Keying (PSK) and Frequency Shift Keying (FSK). Both have their
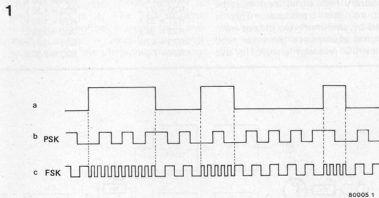
Figure 1. Two possible ways of modulating a digital signal on an ultrasonic carrier. 'a' is the original digital signal; 'b' illustrates the principle of Phase Shift Keying (PSK); 'c' is an example of Frequency Shift Keying (FSK). For PSK, the carrier frequency is unaltered — but its phase is shifted 180° at every zero-crossing in the digital signal; FSK, on the other hand, changes the carrier frequency.
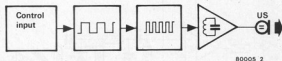
Figure 2. Block diagram of the transmitter.

transmit 'analogue' control signals. The latter can be useful for volume or balance control, brightness or contrast on a TV set, and so on. Perhaps it should be noted that 'on/off' type control signals can also be used for analogue functions, by converting them — at the receiver end — into 'more/less' type control. Opening or closing curtains, for example: by switching a motor on and off at the correct moments, the curtains can be opened as far as you like.

own advantages and disadvantages, but tests have shown that FSK has more disadvantages than FSK. In particular, PSK is more 'vulnerable' to the Doppler effect; an FSK system, on the other hand, can be made virtually immune to this kind of problem.

Having chosen FSK, we can proceed to the block diagram of a suitable transmitter: see figure 2. This contains two square wave generators, the first running at a much lower frequency than the
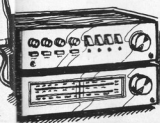
second. The frequency of the first squarewave generator is to depend on which of the sixteen control signals is to be transmitted. This implies that only one command can be given at a time: it would be asking rather a lot for the generator to run at two different frequencies simultaneously! The sixteen different command frequencies are in the 6 Hz to 90 Hz range.

The second square wave generator is controlled by the first: its output frequency is 38.5 kHz when a 'high' voltage is applied to its input, and 40.9 kHz for a 'low' input. The output from this second squarewave generator is transmitted as an ultrasonic wave, by the US transducer. To achieve a reasonable control range, a sufficiently high signal level must be applied to this transducer. This is achieved by amplifying the output from the second squarewave generator, and using an LC resonant circuit. In this way, voltages of up to 150 V can be produced across the transducer, without the need for high supply voltages. The LC resonant circuit has a further advantage: it converts the square wave into something that resembles a sinewave — and most US transducers prefer it that way.

In practice

The transmitter circuit is given in figure 3. As mentioned earlier, sixteen control inputs are provided. When a control signal is to be transmitted, the corresponding control input to the transmitter must be connected to supply common — by means of a pushbutton, say. Only one control input should be selected in this way at any given moment. It may be noted, in passing, that the outputs of the Monoselektor mentioned earlier are also 'active low'. This means that they can be connected direct to the inputs of this transmitter circuit.

When one of the control inputs is connected to supply common, the corresponding transistor (T1 . . . T16) starts to conduct. One of the sixteen resistors R17 . . . R32 then goes into action as a frequency-determining element for the first squarewave generator, IC1. This IC

Table 1.

Resistor values for figure 3.

R17 = 15 k	R25 = 68 k
R18 = 18 k	R26 = 82 k
R19 = 22 k	R27 = 100 k
R20 = 27 k	R28 = 120 k
R21 = 33 k	R29 = 150 k
R22 = 39 k	R30 = 180 k
R23 = 47 k	R31 = 220 k
R24 = 56 k	R32 = 270 k

3

FF = 1/2 IC2 = 1/2 4013

T1 . . . T16 = BC 557
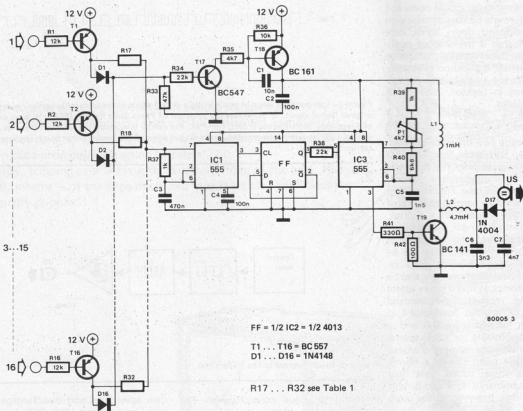D1 . . . D16 = 1N4148

R17 . . . R32 see Table 1

80005 3

Figure 3. Complete circuit of the transmitter.

starts to produce an output signal with a frequency that is determined by the selected control input (the squarewave frequency varies between 12 Hz and 180 Hz for the resistor values given in the Table). The following flipflop, FF, halves this frequency; of more importance is the fact that it produces a nice, symmetrical squarewave output. This signal goes to the modulation input of a second 555 (IC2), that produces the modulated carrier signal (at approximately 40 kHz). A transistor, T19, is used as output amplifier. The various inductors are included to boost the output signal even further and to clean it up — providing a sinewave at some 150 V peak-to-peak across the transducer.

The three ICs and the output stage are not connected directly to the positive supply rail; an electronic switch, consisting of T17 and T18, is used. Sixteen diodes (D1 ... D16) are used in an OR gate configuration, switching on the electronic switch as soon as one of the control inputs is activated. When no control signal is to be transmitted, the electronic switch opens; the circuit then draws virtually no current. A useful feature for a circuit that will almost certainly be battery-powered!

The receiver

The transmitter circuit described above is built into the remote control unit. It transmits sixteen commands as a modulated ultrasonic signal. This signal must be received and 'decoded' at the other end, to control the equipment as required.

The block diagram of the receiver is given in figure 4. The first sections — trigger, monoflop, low-pass filter and another trigger — amplify the signal pciked up by the ultrasonic transducer. This signal is then passed to one or more digital filters; the circuit of one of these

(US) and recover the low-frequency squarewave from it (the 6 Hz ... 90 Hz control signal). This signal is then applied to one or more (up to sixteen) digital band-pass filters that serve to distinguish between the sixteen channels. When a command is transmitted, only the digital filter that is 'tuned' to the corresponding low-frequency squarewave will produce an output. This signal toggles a flip-flop, FF. Giving a command therefore results in this flip-flop changing state; if it, in turn, is used to control a lamp, the lamp can be switched on and off by means of the remote control unit. The whole point of the exercise!

The circuit of the receiver is divided into two sections. The first part (figure 5) receives the ultrasonic signal and retrieves the low-frequency squarewave. This signal is then passed to one or more digital filters; the circuit of one of these
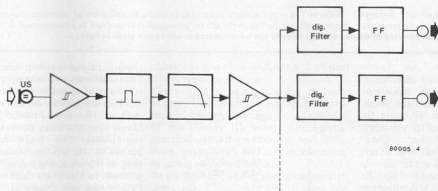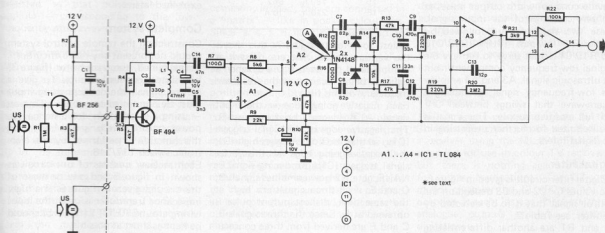

Figure 4. Block diagram of the receiver.

Figure 5. The first part of the receiver circuit, up to the digital filter(s).
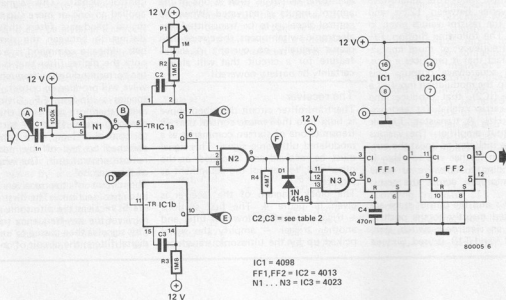
6

Figure 6. The circuit of a digital filter. Up to sixteen of these circuits can be connected to the circuit given in figure 5.

filters is given in figure 6.
Depending on the type of transducer used, either of two possible input circuits may be required. The input stage shown at the upper left in figure 5, using FET T1, is intended for use with high-impedance transducers. Low-impedance transducers, on the other hand, can be connected direct to C2.
The ultrasonic signal is amplified by T2. The resonant circuit (L1/C3) is tuned to 40 kHz. The signal is then 'squared up' by A1 and A2, producing a squarewave that switches between supply common and the full supply voltage. This signal goes to a complicated network around A3. In brief, R12-C7-R14-D1 and R16-C8-R15-D2 each operate as simple differentiating networks; they each provide constant-width output pulses at the zero-crossings of the input signal. These two networks are followed by simple RC low-pass filters (R13/C10 and R15/C11) that serve to retrieve the original low-frequency component from the ultrasonic signal. A3 and A4 amplify this low-frequency signal, producing a square-wave that swings between 0 V and full positive supply. This signal is ideally suited for further processing in the digital filters.

Digital filter

A digital filter circuit is given in figure 6. The values for C2 and C3 depend on the control signal that is to be detected by the filter; see table 2.
C1 and R1 are another differentiating network; together with N1, they produce pulses with constant width. IC1a and IC1b are monostable multivibrators. To understand the circuit, it is

Table 2.

Values for C2 and C3 in figure 6.

control signal	C2	C3
1	10 n	470 p
2	15 n	470 p
3	15 n	820 p
4	18 n	820 p
5	22 n	1 n
6	33 n	1 n
7	39 n	1n5
8	47 n	1n5
9	56 n	2n2
10	68 n	3n3
11	82 n	3n3
12	100 n	4n7
13	120 n	4n7
14	150 n	6n8
15	180 n	8n2
16	220 n	10 n

easiest to refer to figure 7: this shows the various signals that are present at the indicated points in figure 6, when a signal is being received at the 'resonant frequency' of this filter.
Each negative-going edge in the input signal, A, produces a brief pulse at B. The negative edge of this signal triggers IC1a, so that its Q output goes high (D); the negative-going edge of this latter signal triggers IC1b, producing signal E. NAND gate N2 now combines signals B, C and E; if all three signals are 'high' at the same time, a short output pulse is obtained at F. Since the three signals B, C and E are derived from three consecutive input pulses, with delays determined by the monostable time constants, an output signal at F can only be produced if the frequency of

the input signal is almost exactly right.
The brief negative output pulses from N2 are used to clock flip-flop FF1. This flip-flop is actually used as a monostable multivibrator, with a period time that is longer than the delay between consecutive pulses. The result is that the output of this flip-flop remains 'low' as long as N2 is being clocked by pulses. This signal is used to clock the second flip-flop, FF2 — the one shown as 'FF' in the block diagram given in figure 4. The complete circuit therefore works as a kind of 'selective flip-flop': it is not toggled by a single pulse; instead, it responds to a square-wave input at exactly the correct frequency.
Preset P1 is used for setting the 'resonant frequency' of the digital filter, as will be explained later.

Complete system

Constructing the remote control system should not present any real difficulties. As explained, the transmitter (figure 3) has sixteen control inputs. To give a control signal, the corresponding input must be connected to supply common — using a pushbutton, for instance. No power on/off switch is required, since this function is performed by the electronic switch T17/T18.
Each receiver consists of one circuit as shown in figure 5 and one or more of the circuits given in figure 6. If a high-impedance transducer is used, the input wiring around FET T1 (figure 5) should be kept as short as possible.
When setting up the transmitter circuit, the signal at the output of A2 in the receiver ('A' in figure 5) must be monitored. This signal can be measured, of
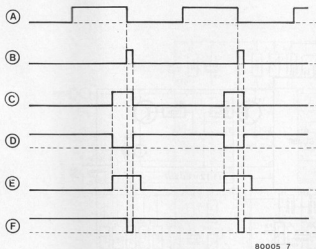
7

8

80005 8

Figure 7. Six signals, occurring at the indicated points in the circuit given in figure 6. The input squarewave, A, is assumed to be at the 'resonant frequency' of the filter, so that negative output pulses occur at F.

Figure 8. This little circuit is used to adjust the remote control system. A high-impedance headphone should be used.

course, but it is just as easy to 'listen' to it. A high-impedance headphone is used, with a series resistor and capacitor, as shown in figure 8. This little circuit is connected between point A in the receiver and supply common. One of the control inputs to the transmitter is activated, and P1 in the transmitter is adjusted until a tone is heard in the headphone. This tone should be present over a small part of the adjustment range of P1; the preset is now set to the centre of this range.

The same headphone circuit can be used to adjust the digital filters. It is connected to the output of N2 ('F') in figure 6. The control input to the transmitter that is to correspond to this filter is activated, and P1 in the filter is adjusted so that pulses are clearly audible — corresponding to the low-frequency squarewave.

The easy way out: two ICs

With all its advantages, the remote control system described above would never win an award for low component count. Obviously, manufacturers of TV receivers would prefer a more compact system, and IC manufacturers have been quick to meet the demand. One special IC in the transmitter and one in the receiver takes a lot of hard work out of the manufacturing process.

One manufacturer who supplies a pair of ICs for this job is Plessey. The SL490 (transmitter) and ML920 (receiver) form the main components in the second remote control system that we will describe. These ICs can be used in either infrared or ultrasonic remote control systems. For several reasons, an

ultrasonic system is preferable — and so that is what we will use.

There is little point in going to such extensive detail on the operation of this twin-IC system. Discussing the 'innards' of the Plessey ICs would be of little practical use — there's no way to modify them. Instead, we will restrict ourselves to giving suitable circuits.

As we shall see, one of the advantages of these ICs is that they can be used for transmitting 'analogue' signals as well as the digital kind.

A transmitter circuit using the SL490 is given in figure 9. As in the previous system, the control signals can be given by means of pushbuttons. However, in this case they are not connected between a control input and supply common; instead, they are connected in a 'matrix' — like in pocket calculators. Since a 4 x 8 matrix is used, up to 32 pushbuttons can be used.

Of the 32 pushbuttons (assuming that all should be operated . . .) only one should be operated at a time. The position of this button is converted into a five-bit binary code, in the IC. The code numbers run from 00000 (upper left) to 11111 (lower right); they increase progressively from left to right and then down to the next row — like reading lines on a page. In effect, this means that the two right-hand bits in the code are determined by the position in the row (horizontally), whereas the other three bits indicate which row is meant.

This five-bit code is transmitted, using a special type of modulation: pulse position modulation, or PPM. A series of six pulses of equal length are trans-
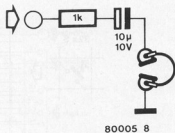
mitted; the five periods between the pulses can be either long or short, depending on the five-bit code. A long 'pause' corresponds to a logic 0, and a short interval is for a logic 1. This is illustrated in figure 10. Preset P1 is used to adjust the correct pulse/pause lengths. The interval between pulses should be approximately 20 ms for a logic 1 and 30 ms for a logic 0; the pulse length is approximately 10 ms. The ultrasonic carrier is transmitted during the pulse. The carrier frequency is set by P2; a good choice is 40 kHz.

A few transistors are added to boost the current through the transducer — the IC itself can supply only 5 mA. The maximum range of the system is increased in this way. As in the system described earlier, this one also contains an electronic supply switch — included in the IC — so that the current consumption from the 9 V battery is only 6 μA if no command is being transmitted.

The receiver

The ultrasonic commands given by the SL 490 transmitter can be decoded by a receiver using the ML 920. This IC is intended for use in colour TV receivers, but there is nothing against using it for other applications. The 32 commands given by the SL 490 are not all decoded as such; instead, there are two analogue outputs as well as several digital ones.

Before taking a closer look at what the ML 920 can do — in conjunction with the SL 490 — we should first point out what is cannot do: amplify and de-
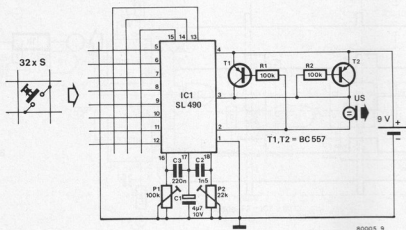
9

Figure 9. The 'fully integrated' transmitter.

modulate the ultrasonic signal. This IC expects to receive a clean series of pulses, like those shown in figure 10, without any ultrasonic components. A separate preamplifier and demodulator must therefore be included between the transducer and the IC. No problem, as we will see.

The circuit that distills the commands from the 'clean' pulse signal is given in figure 11. Not counting a 'monitor' output for adjustment purposes, there are thirteen 'true' outputs. These are divided into three groups: three analogue outputs (A1 ... A3), five digital outputs (D1 ... D5) and five 'channel' outputs (C1 ... C5).

The 'channel' outputs are intended for switching channels on the TV-receiver. Up to twenty channels can be selected by these five outputs: the desired channel number (0 ... 19) is available as a binary five-bit code (00000 ... 10011) at outputs C1...C5. When one of the commands 00000 ... 10011 is given from the control unit, the channel output at the channel outputs assumes the same value. Whenever the value at the C outputs changes (when a channel change is required) a short pulse also appears at digital output D4; this can be used for various applications. If a command with a higher number than 10011 is given by the control unit, the channel outputs remain unchanged and no pulse appears at the D4 output.

It is also possible to step through the

10

Figure 10. The IC transmits the five-bit code in a chain of six pulses (two of the pulse trains are shown). The information is contained in the interval between pulses: a logic 1 gives a short pulse, logic 0 is coded as a longer interval. Both pulse trains shown here give the same code.

channels automatically. Control signal 10101 causes the receiver to step through the channels in ascending order; code 11101 makes it step through the channels in the opposite direction. If, in some application, the twenty channels are required individually — instead of as a five-bit code — they can be 'demultiplexed' by means of a suitable IC (the CMOS type 4514, for instance). The only point to note, in that case, is that the ML 920 uses 'negative logic': logic 0 is positive supply voltage, logic 1 is equivalent to 0 V.

The analogue outputs of the ML 920 are intended for controlling volume, brightness and colour saturation. The voltages at these outputs vary in sixteen steps. Their functions, and the corre-

Table 3.

Functions and control signals for the analogue outputs in figure 11.

output	increase	decrease	function
A1	10100	11100	colour saturation
A2	10110	11110	volume
A3	10111	11111	brightness

sponding control signals, are listed in Table 3.

The digital output D2 is intended for controlling the 'colour killer': it provides a logic 0 (almost positive supply level) when the analogue colour output A1 is at zero level — this corresponds to 'no colour'. A 'mute' command is also provided, for temporarily switching off the sound: the control signal 11001 switches output D1 alternatively low and high. Output D3 provides the 'on/off' signal for the TV receiver; the corresponding control signal is 11000. More accurately, this should be referred to as a 'stand-by' switch — obviously, the TV receiver cannot be switched off altogether: at the very least, the ultrasonic receiver must be left on.
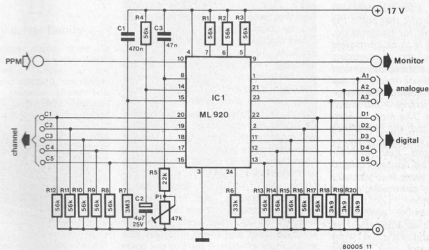
11

Figure 11. This circuit decodes the pulse trains, producing a series of digital and analogue outputs intended for controlling various functions in a colour television receiver.
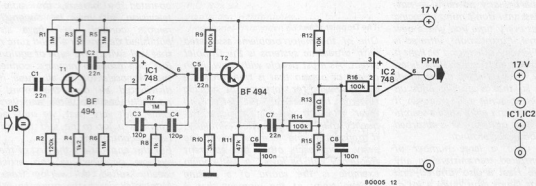

Figure 12. This preamplifier and demodulator must be included between the transducer and the circuit given in figure 11.

Finally, a 'basic setting' command (11011) causes all three analogue outputs to assume an intermediate level — approximately one-third of their range.

There is only one preset adjustment in the ML 920 circuit: P1. This must be adjusted so that a squarewave appears at the monitor output (pin 9) with a period time that is one-twentieth of the interval time for a logic 0 in the PPM signal.

As stated earlier, the transducer cannot be connected direct to the IC. A preamplifier is required, that also removes the 40 kHz components from the signal. A suitable circuit is given in figure 12. The ultrasonic signal is amplified by T1. IC1, with the associated components, is used as a 40 kHz band-pass filter. The main reason for including this is to reject the second and third harmonics

of the TV line frequency (31.3 kHz and 46.9 kHz).

Demodulating the ultrasonic signal is no problem — it calls for nothing more than filtering out the 40 kHz component from the signal. This is done by means of a low-pass filter (IC2).

The output of the circuit given in figure 12 can be connected direct to the input of that in figure 11.

Wireless: three varieties

For 'wireless' transmission of control signals or other data, a few different options are available.

The most common system is radio — in fact, 'wireless' has become almost synonymous with this. The control unit contains all controls, a (miniature) radio transmitter, and a circuit that codes the various control signals for transmission. The equipment that is to be controlled contains a receiver and a decoding circuit that retrieves the original control signals. A radio link of this kind has several advantages — considerable range for low power, for instance — but it also has its disadvantages. The most important 'disadvantage' is that the Post Office authorities take a very dim view of people operating this kind of link without a licence. This limits its uses to exactly those fields were it is virtually the only possibility — controlling model aeroplanes, for instance.

For use indoors, remote control by radio is 'not on'. As an alternative, we could consider using light. A light source in the control unit, a light detector in the receiver, and suitable coding and decoding circuits are again required. To avoid undesirable 'disco lights' effects, it is now common practice to use light that is invisible to the human eye: 'redder than red' — infrared, in other words. This has a further advantage in that it penetrates smoke and dust haze better than visible light.

However, an infrared remote control system is not an ideal solution. It is no easy matter to make the detector in the receiver sufficiently sensitive. Nor can the infrared light source in the transmitter be particularly powerful — not, that is, unless you don't mind changing the batteries every time you give a new control signal. Furthermore, infrared is light; and like any other kind of light it tends to travel in very straight lines. Admittedly, it is reflected off all kinds of objects so that it usually ends up at the receiver some time — even if the transmitter isn't aimed like a searchlight — but a lot of its power is absorbed by all that bouncing around.

There are also a large number of 'spurious infrared transmitters' in any living room. Heat is also infrared, and incandescent lamps emit quite a lot of energy in the infrared region that we are interested in. What all this means is that only a little of the desired infrared signal will reach the receiver, and that there is a lot of interference. For this reason, both the transmitter and the receiver circuits must incorporate all kinds of sophisticated tricks, if the receiver is to be able to distinguish the control signals from the 'noise'.

A sound system

We have seen that neither radio nor infrared links are ideal for domestic use. The former is forbidden, and the latter doesn't work very well — at least, it takes some doing to make an infrared system work reliably.

Fortunately, a third option is available: sound waves. For use indoors, these are much more suitable. In particular, it is much easier to 'fill' a room with sound: the waves bounce and bend all over the place. This is still true at the relatively high frequencies that are of interest for remote control systems.

When discussing light as a transmission medium, it was pointed out that 'invisible' light is preferable to the visible variety, to avoid being dazzled by the light show put out by the control unit. For the same reason, 'inaudible' sound is preferable to the normal variety in this application. By inaudible, we mean above the range of human hearing (maybe a few dogs or bats may hear these control signals): 'ultrasonic', as it is called.

Using high frequency sound has a further advantage: the 'loudspeakers' and 'microphones' (both are usually referred to as 'transducers', actually) are both small and cheap. They are also surprisingly efficient — putting out quite a lot of 'sound' for a fairly modest electrical power consumption.

So why doesn't everybody use ultrasonic transducers for remote control systems? They must have some disadvantages? Rest assured, they have. They are more bulky than infrared LEDs; they are more 'breakable'; and 'spurious ultrasonic transmitters' also exist. Rattling keys, for instance. For this reason, a sufficiently 'interference-proof' coding and decoding system is still required.

There is one further problem:

The Doppler effect

One of the major problems associated with ultrasonic systems is the Doppler effect. As most people will know, if an object or person that is transmitting a sound wave, for instance, is moving towards the receiver, the latter will 'hear' a sound wave at a higher frequency than that transmitted. Similarly, if transmitter and receiver are moving away from each other the apparent frequency will be lower. A well-known example is the sound of a passing express train: at the moment that it passes us, the sound seems to suddenly drop to a much lower frequency. Obviously, the train doesn't really 'change its tune' when it sees us. The frequency that we are hearing is not the same as that which the train is 'transmitting'.

Indoors, it is unlikely that we will travel at the speed of an express train. Even so, the Doppler effect can be quite a nuisance when using an ultrasonic remote control link. (The same applies to infrared links, by the way, but to a lesser extent.)

When designing a system, all due care must be taken to ensure that it is relatively insensitive to the kind of frequency shifts that can be caused by this effect. Fortunately, this is not too difficult. The two designs given in this article are 'Doppler-proofed'.
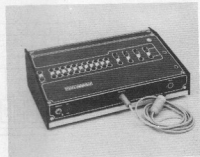
The Monoselektor goes remote

The Monoselektor described in Elektor, June 1979, can be used to control a large number of different things at the touch of a single button: radio and/or television set, lights, doors, curtains, and so on.

There is no problem in constructing the unit, as experience has shown, but when it comes to putting it into operation things are apt to get 'ropey'. Or 'wirey', rather. The curtains and door may be operated by motors, the radio and television set may be designed for remote control; we have already published designs for a 'solid state relay' that will switch this type of equipment. But one problem remains: connecting the Monoselektor to all these items, distributed all over the room. This normally involves yards and yards of cable. Not very neat, we must admit. There is a lot to be said for using a remote control link.

For this application, the first of the two systems described in this article is ideally suited. All wiring from the Monoselektor to the other equipment can be replaced by a 'wireless' link. The ultrasonic transmitter (figure 3) is built into the Monoselektor case; the control inputs can be connected direct to the Monoselektor outputs.

Receivers (figures 5 and 6) can be mounted wherever required in the room, to control the motors, relays or whatever. The fact that several receivers can be used in the same system is an advantage. In fact, if cost is no problem, sixteen receivers can be built, each tuned to its own channel. In practice, of course, it will normally be more economical to use one receiver for several items of equipment that are close to each other in the room.

market walker

Versatile display driver family

A new series of dot/bar display driver ICs have been announced by National Semiconductor Corporation. Known as the LM3914, LM3915 and LM3916, the devices provide a highly readable all-electronic display. The LM3914 family can be cascaded without needing individual adjustment and production trimming.
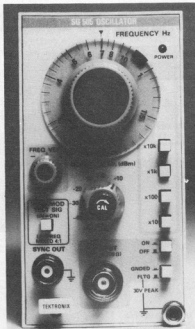
Capable of driving LEDs, vacuum fluorescent, or even liquid crystal displays, these devices can provide a linear, logarithmic or VU readout. The LM3914 is a ten-step linear display driver, while the LM3915 covers a 30 dB range with ten 3 dB steps, and the LM3916 covers the familiar + 3 VU to −20 VU range. The LM3914 is a large scale linear integrated circuit containing 10 precision comparators with a divider to give millivolt accuracy at each trip point, a built-in voltage reference, plus bar-dot logic circuitry and constant current output drivers. The family can be operated from power supply voltages between 3 V and 20 V, and is easily scaled to accept signals from millivolts to volts. In addition, the devices are protected from input overvoltage up to ± 35 V. The output current from open-collector transistors can be programmed from less than 1 mA to over 20 mA allowing the device to be used even with high impedance sources.

Another advantage is the ease with which these devices can be expanded to 50 or more outputs for high resolution displays.

The LM3914/LM3915/LM3916 series is available in an 18 pin dil package.

National Semiconductor (U.K.) Ltd.,
Telephone: Bedford (0234) 47147

(1449 M)

Low distortion audio oscillator

The SG 505 audio oscillator is a new plug-in unit for the Tektronix TM500 family of modular test and measurement instruments. The oscillator covers the range 10 Hz to 100 kHz and its total harmonic distortion between 20 Hz and 20 kHz is less than 0.0008%. It has four overlapping frequency ranges which are selected by push-button controls. Within each range, the required frequency is selected by a continuous tuning dial which indicates the exact frequency within a tolerance of ± 3%. A vernier knob with a ± 1% range simplifies fine frequency adjustments.

Output signal amplitude is greater than 6 V rms unloaded and greater than 3.12 V rms into 600 Ω. Control of the output is provided by a precision 8-step attenuator ranged from +10 dBm to −60 dBm. Step accuracy for each 10 dB increment is ± 0.1 dB. A variable control allows for continuous adjustment within each step. Level flatness (referenced to 1 kHz) is ± 0.1 dB from 20 Hz to 20 kHz and ± 0.12 dB above this range. An on/off switch allows the output signal to be disconnected whilst a second button selects between ground refer-

enced of floating output. A sync output terminal provides a 200 mV rms sinewave which tracks the selected output frequency. Both the output and the sync signals are available at the rear connector of the SG505 as well as at the front panel. An optional intermodulation test signal feature provides a combined signal consisting of a low frequency sinewave (60 Hz or 250 Hz) and the dialselected frequency. These two components are mixed in a 4:1 amplitude ratio, the amplitude of the combined signal is within + 0.2 dB of the amplitude of the fundamental signal alone.

Tektronix UK Ltd., Beaverton House,
P.O. Box 69,
Harpenden, Hertfordshire.
Telephone: Harpenden 63141.

(1454 M)

Dual-in-line converters

A new series of 1 Watt DC to DC converters has been introduced by Gresham Lion Limited.

FUNCTION		DUAL VRE
Input	12.14	
Input	12.14	
Output	10.15	

The new models come in standard 24-pin DIL packages and will be known as the Gemini 600 series.

As these units will fit into standard IC sockets and printed circuit board drilling patterns, on-board power conversion systems may be assembled quickly and economically. The devices are suitable for automatic insertion techniques, and may be flow-soldered after insertion.

Although the units are small, they have all the features of much larger modules. They are available with single and dual outputs, have output regulation and are fully isolated and short circuit protected. Initially, input voltages are either 5 V or 12 V nominal, with five possible output configurations for each input including 5 V, 12 V, 15 V, + 12 V and ± 15 V (all 5%).

High reliability is claimed for the units — a minimum MTBF of 100,000 hours — and the operating temperature range is −25°C to +70°C. Load regulation is only 150 mV from zero to full load and the output ripple is a maximum of 30 mV p-p. Output current limiting ensures that the short circuit current is only 150% of the full load current.

Units from the Gemini 600 series measure only 1.28 x 0.6 x 0.55 inches overall (LxWxH). Pins are on standard DIL 0.1 inch spacing, 0.6 inch pitch.

Gresham Lion Limited, Gresham House,
Twickenham Road, Feltham,
Middlesex, TW13 6HA,
Telephone: 01-894 5511.

(1453 M)

New size of Verocase

A new size of vacuum-formed Verocase has been introduced by Vero Electronics Limited to compliment the existing range. Although specially designed for housing a keyboard and display panels, this enclosure has wide ranging applications in the instrument field.
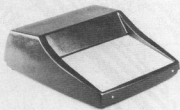
Vacuum-formed from black textured ABS, the case is supplied in two sections which screw together. The material may be easily cut or drilled as required for switches, displays etc. A flat area is provided at the rear of the case for cable entry.

This intermediate size, with outside dimensions of 340 mm wide x 360 mm deep x 130 mm high is supplied with an anodised aluminium front panel 254 x 170 x 2 mm thick, four self-adhesive rubber feet and case assembly screws.
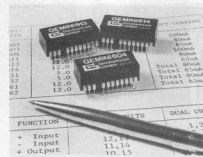
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(1452 M)

(1452 M)